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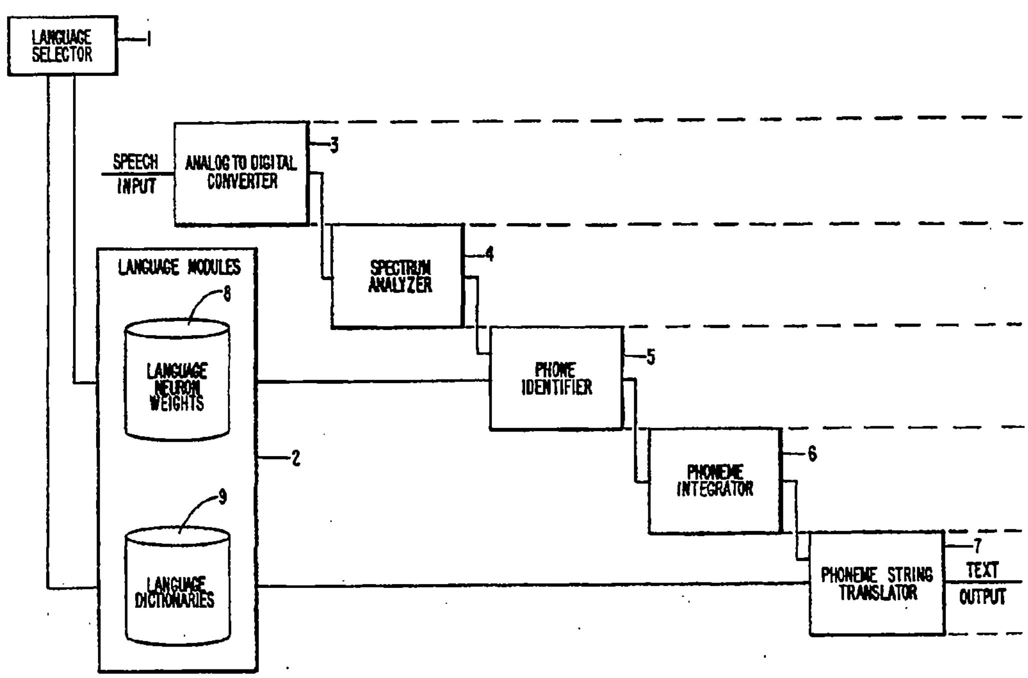

(57) Abstract

Speech input is converted to digital data (3) and undergoes spectral analysis (4). The spectrum is analyzed to identify phones (5) using a standard neural network implemented using stored weights (8). The phones are further combined to identify phonemes (6). The phonemes are then translated into a different language (7) based on a stored language dictionary (9) and converted to text output.

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DESCRIPTION

Multi-Language Speech Recognition System

Field of Invention

The present invention relates to speech recognition systems and methods.

Background

The prior art includes many systems and methods for transcribing speech. One of the major differences between them is the level of difficulty of the speech recognition task they are intended to perform. The simplest such task is the recognition of a small number of acoustically distinct words spoken in isolation (often called discrete speech). U.S. Pat. No. 4,910,784 to Doddington et al ("Low Cost Speech Recognition System and Method") is an example of the prior art of this class of system. Such systems are useful, for example, for giving a small set of commands to operate a computer, but can not handle continuous speech. A more difficult type task is the identification of one or more designated words occurring in a continuous stream of words or "word spotting". U.S. Pat. No. 4,937,870 to Bossemeyer, Jr. is an example of the prior art of this class of system. Such systems might be used, for example, in a telephone application for identifying key words or phrases within an utterance such as "credit card", "collect", "third party", etc. but can not transcribe continuous speech. A still more difficult type task is the recognition of all words in a complete sentence where the words are spoken in isolation and the grammatical structure of the sentence is prescribed. U.S. Pat. No. 4,882,757 to Fisher et al ("Speech Recognition System") is an example of the prior art of this class of system. Such systems can be useful in applications where the speaker is willing to accept speaking in an un-natural manner to accommodate the needs of the system. An even

more difficult speech recognition task is the recognition of all words in a complete sentence when the words are connected (generally referred to as continuous speech), the grammatical structure of the sentence is prescribed and the lexicon is constrained. U.S. Pat. No. 5,040,127 to Gerson ("Continuous Speech Recognition System") is an example of the prior art of this class of system. Such systems can be useful in task-specific applications where the user is aware of the system vocabulary and grammar constraints and able to modify his or her speech pattern accordingly. The most difficult type of task is the recognition of all words in a continuous, spontaneous utterance that may have no structure and indeed may be ungrammatical in form. U.S. Pat. No. 4,852,170 to Bordeaux ("Real Time Computer Speech Recognition System ") is an example of the prior art of this class of system.

Systems and methods of speech recognition may also be classified according to whether they are speaker-dependent; i.e., must be trained by a particular speaker prior to that speaker making use of the machine, or whether they are speaker-independent; i.e., a particular speaker need not train the machine prior to using it. A variation of the speaker-dependent type is the speaker-adaptive system which aims to make the training of the machine easier and faster. Speaker-independent systems are more difficult to achieve than are speaker-dependent ones; however, in most applications they have much greater utility. The present system described herein is speaker-independent.

Speech recognition systems and methods further may be classified as to the lowest phonetic unit that they identify. Every system is provided a set of spectral reference patterns for each of the lowest phonetic units to which the incoming speech signal is compared to seek a best match for identification. The largest such unit is a whole word (or small group of words). Systems operating with reasonable accuracy at this phonetic level generally

are limited to small-vocabulary, discrete-speech applications. Methods which aim to identify phonemes for aggregation into words are represented in the prior art across classes of speaker-dependent and -independent/discrete- and continuous- speech systems. Difficulty in achieving reliability is encountered in such systems as larger vocabularies introduce more similar sounding words and as multiple speakers introduce different pronunciations of the same words. Methods of identifying phones (i.e., sub-phoneme units of speech) aim to achieve improved reliability by identifying more but smaller segments of the speech signals. The present system described herein includes a method for accurately identifying phones.

Speech recognition systems and methods may be still further classified according to their modelling of the speech process. Some methods describe the process as a series of acoustic events. This model has been applied primarily to phoneme recognition. In such a model, the speech signal is first segmented into occurrence of classes of phonemes such as vowels (/IY/, /OW/, etc.), fricatives (/F/, /S/,etc.), stops (/D/, /T/, etc.) and so forth. Then the specific phoneme within the class is identified. A second model takes the view that it is not possible to analyze the speech process directly but that it can be usefully analyzed in statistical terms. The Hidden Markov Model is an example of this view of the speech process. In this model, segments of the speech signal are considered as (spectral) states of the system and transitions from one state to any other are probabilistic. Each phoneme or phone is described in terms of a sequence of state changes. The probabilities of transition between spectral states of an incoming speech signal are calculated to determine probable correspondence to each of the target sequences to determine probable phoneme or phone identification. It is difficult to achieve high reliability with this method in

large vocabulary speaker-independent systems because of the much larger number of possible spectral states compared to the number of spectral states in a speaker-dependent system. A third model views the speech signal as a sequence of spectral patterns; i.e., a directly observable representation of the signal. This is the model that is employed in the present invention as will be described in detail later.

All speech recognition methods are based on comparing the characteristics of the unknown speech signal with a reference set of examples to determine when a "good" match occurs (identification). Thus, another way of classifying speech recognition systems and methods is on the basis of how the reference data is derived to which the unknown speech signal is compared for identifying a word, phoneme or phone. In a "rules are given" system, the system designer provides the machine directly the reference data to be used for determining best matches. The designer devises the shapes of the templates or calculates the state transition probabilities as in a Hidden Markov Model approach. Speaker-independent applications result in a need for a large number of spectral states to accommodate the wide variations in speaker's voices. Spectral states that are similar may be aggregated but at some loss of representational accuracy and hence reliability of identification. In a "rules are learned" system (e.g., an artificial neural network), the designer provides the system with a very large number of examples of spectra of each phone of interest and their identification. The system is run in a training mode and the neural network "learns" how to distinguish one phone from all the others. When run in an application, the neural network determines the probability that the segment of signal encountered is each of the phone possibilities. Selection is made when specified probability threshold criteria are met. This is the method used in the present invention. An important advantage of this approach in speaker-independent

applications is that its reliability can be improved with the number of speakers using it.

A final way of classifying speech recognition systems relates to the aids to word identification employed, if any. In a "context-free" scheme, the string of phones or phonemes are compared to lexicon or dictionary entries to identify each word directly. In a "context-assisted" scheme, devices such as allowable word pairs, constrained grammar and/or syntax and the like are used to improve reliability of word identification. The present invention is context-free.

Most of the speech recognition methods described by the prior art can be modified for application to other languages. However, those methods that depend on analytical devices such as allowable word order, grammar and/or syntax to assist in word identification require separate and duplicative effort for cross-language implementation. In an era of global communication and commerce, there is a need for a language-independent system that has not heretofore been addressed by the prior art. The design and implementation of such a system will exploit the overlap in the speech sounds used in different languages. Exploitation of the common usage of sounds between languages requires application of a more detailed understanding of speech production and the resultant speech signal than has been the case in the prior art.

Summary of the Invention

The prior art has not taught the construction of devices with the capability to mimic human capability to recognize phones; i.e., "a speech sound considered as a physical event without regard to its place in the sound system of a language." (Webster's Ninth New Collegiate Dictionary; Merriam-Webster Inc., Publishers; Springfield, MA; 1991) "Human languages display a wide variety of sounds, called phones or speech sounds. There are a great many speech sounds, but not an infinite number of

them....The class of possible speech sounds is not only finite, it is universal. A portion of the total set will be found in the inventory of any human language." (Contemporary Linguistics: An Introduction; William O'Grady, Michael Dobrovolsky, Mark Aronoff; St. Martin's Press; New York; 1989).

It is an object of my invention to provide a system and method for recognizing the total set of speech sounds (or phones) in human languages.

It is another object of my invention to provide a system and method for transcribing the speech of arbitrary speakers in one of many languages including when such speech is continuous and conversational.

It is yet another object of the present invention to provide a system and method for processing the speech signal to yield an accurate determination of all the frequencies contained in that signal and their amplitudes.

It is a further object of the present invention to emulate the human hearing processes to provide a system and method for unique direct observation of the perceived speech signal at very short time intervals.

It is yet another object of the present invention to address the phones in a language as fuzzy sets; i.e., as all speech signals having a probabilistic membership in all phone sets.

It is a still further object of the invention to provide an artificial neural network system and method for determining the probable phone represented during each very short time interval.

It is a further object of the invention to provide a unique method of employing the artificial neural network to identify the time during the utterance of a phone which represents the closest approach of the vocal tract configuration to a target position; i.e., when there is the maximum likelihood of the signal representing the intended phone.

It is another object of the invention to provide a method for accommodating multiple pronunciations of the same word.

It is yet another object of the invention to provide a method of separating words that often are run together in conversational speech due to coarticulation.

It is still another object of the invention to provide a method of exploiting the common usage of some phones between languages so that the inclusion of other languages is efficiently accomplished with the time required for each new language decreasing with the number of languages included.

Exploitation of the common usage of sounds between languages requires application of a more detailed representation of speech production, the resultant coding of the speech signal, and emulation of the neurophysiological mechanisms of hearing and pattern recognition that decode that signal to allow speech recognition than has been the case in the prior art. The present invention emulates the concurrent processes occurring in humans recognizing speech; i.e., spectrum analysis, speech sound identification and word recognition. The frequency response and sensitivity of human hearing is mimicked, an artificial neural network is included to represent the pattern recognition apparatus of the brain and logical processes are included to emulate our translation of spoken sounds into written words.

These and other objects and features of the present invention will be better understood through a consideration of the following description taken with the drawings in which:

Figure 1 is a logical diagram of the system.

Figures 2A-2C are illustrations of a simplified source-filter decomposition of a voiced sound. Figure 2A is a typical source spectrum, 2B is a representative vocal transmission filter function, and 2C is a spectrum of a radiated vowel.

Figure 3 is a graph of frequency discrimination versus frequency and loudness of a tone.

Figure 4 shows the relative response of some narrow bandpass filters versus frequency.

Figure 5 shows frequency response of human hearing in terms of the intensity of various frequencies required to produce the same perceived loudness.

Figures 6A-6C illustrates three different concepts of speech segmentation.

Figures 7A-7E show some estimated articulatory positions assumed during pronunciation of the word "caw". Figure 7A is the articulatory position for the phoneme /K/, 7E is the position for the /AO/, and 7B, 7C, and 7D are some estimated transition positions between the two.

Figure 8A presents some typical high resolution spectra for the vowel /AH/ and Figure 8B presents some spectra for the vowel /OW/.

Figure 9 is a schematic drawing of an artificial neural network phone identifier.

Figure 10A-10C are a high resolution spectrograms for a particular word sample spoken by a particular speaker shown in three parts for convenience.

Figure 11a through 11d show a sample output of an artificial neural network phone identifier.

Figure 12 is a logical diagram of the phonemic-to-spoken language translation program.

Figures 13A-13B illustrate an implementation of the present invention on a currently available microcomputer. Figure 13A is a side view of the computer and Figure 13B is a rear view of the computer.

Description of the Invention

Figure 1 is a logical diagram of the system. It includes a language selector 1, language modules 2 stored in non-volatile memory and concurrent processors 3-7 each of which operates on the transformation of the speech signal provided by the previous process. Each language

module 2 comprises, for a pre-determined language, the weights for the neural network 8 to be solved for each interval of time and a language dictionary 9 containing the phonemic-to-spoken language translation of the vocabulary words provided. At start-up, the language selector 1 displays a menu of stored languages from which the user selects the one of interest. It retrieves from storage and passes the neural network parameters and weights for that language to the neural network phone identifier 5 and the appropriate language dictionary 9 to the phoneme string translator 7.

Continuous speech signals then are input into a conventional analog-to-digital converter 3 and thence to the spectrum analyzer 4 which operates on the digitized signal concurrently with the analog-to-digital converter processing subsequent signals. The spectrum analyzer 4 is itself a parallel processor as will be described in detail later. The output of the spectrum analyzer 4 is sent to the neural network phone identifier 5 where a phoneme, allophone or other legitimate speech sound in the language is identified (if a phoneme, allophone or other legitimate speech sound is present). This operation takes place concurrently with the analog-to-digital converter 3 and the spectrum analyzer 4 processing further subsequent speech signals. The output of the neural network phone identifier 5 is passed to the phoneme integrator 6 where various tests are made to ensure that real phonemes, allophones and other legitimate speech sounds in the language are separated from fleeting transitions between them and to combine the allophones and other legitimate speech sounds into phonemes. As before, the phoneme integrator 6 is operating on its portion of the speech signal concurrently with the neural network phone identifier 5, the spectrum analyzer 4 and the analog-to-digital converter 3 processing later incoming portions of the speech signal. As the integration of each phoneme is completed, it is sent to the phoneme string translator 7

where it is added onto the end of the existing phoneme string. When there are a pre-determined minimum number of phonemes in the string, the phoneme string translator 7 accesses the language dictionary 9 to parse the string into the words spoken in the speech stream. Each of the parts of the system now will be described in detail.

Analog-To-Digital Converter

A speech signal is input from a source such as telephone, microphone or tape recorder and is digitized by an analog-to-digital converter 3. In the preferred embodiment, the speech recognition system disclosed herein digitizes the incoming signal at 8 KHz and incorporates an anti-aliasing lowpass filter whose response is approximately 60db down at 4000 Hz from its response from 0 to 3800 Hz. In accordance with the current art, the lowpass filter may be of the analog variety operating on the input signal prior to digitization or a digital filter applied after digitization of the analog signal. The output of the lowpass filter is passed to the spectrum analyzer 4.

Spectrum Analyzer

Before describing the spectrum analyzer 4, it is important to consider the nature of the signal to be analyzed. Fant in his book "Acoustic Theory of Speech Production" (Gunnar Fant; Mouton and Company; The Hague, The Netherlands; 1960) described the spectrum of a radiated speech sound as the product of a source spectrum and a vocal transmission filter function as shown in Figures 2A-2C. The source spectrum is the result of the vibrating vocal cords producing a fundamental frequency and its harmonics which decline in amplitude at 6db per octave. The fundamental frequency can range from a low of about 60 Hz for a man with a bass voice to almost 400 Hz for a child. The "filter function" results from the shaping of the vocal tract to produce a particular speech

sound. In prior art utilizing linear predictive coding to describe a speech sound, the object of investigation has been the filter function. However, the ear receives the entire radiated speech sound, not just the filter function. The linear predictive coding process both distorts the speech signal and discards some of the information it contains. The present invention employs an artificial neural network to identify speech sounds; therefore, it was considered advantageous to retain as much signal information as possible by emulating the human hearing process.

A number of approaches have been utilized in the prior art to simulate human response to speech sounds for example as in Pat. No. 4,905,285 to Allen et al ("Analysis Arrangement Based on a Model of Human Neural Responses") and Pat. No. 4,436,844 to Lyon ("Method and Apparatus for Simulating Aural Response Information"). In both these examples of the prior art, the aim is to simulate the output of the cochlea. The present invention addresses the problem not solely as one of simulating the output of the cochlea but one of further representing the speech signal as it is perceived by the brain. For this purpose, it is necessary to provide an arrangement of pseudo-hair cells providing both the frequency discrimination capability and frequency response of human hearing as determined by auditory testing. The results of one such set of tests of frequency discrimination are illustrated in Figure 3 from "Hearing, Taste and Smell"; by Philip Whitfield and D. M. Stoddard; Torstar Books; New York; 1985. Figure 3 shows that human ability to discriminate between two closely spaced tones is dependent on both the amplitude and frequency of the signal. Higher frequency tones must be spaced further apart for discrimination and higher amplitude ones can be discriminated better than can lower ones.

In order to obtain a representation of the radiated spectrum of the speech signal comparable to human auditory

perception, the preferred embodiment of the present invention employs a plurality of very narrow bandpass filters spaced from 58 to 3800 Hz according to the 10db sound level (upper) curve of Figure 3. Some people with very good hearing still have good speech perception at this signal level. This results in a bank of 420 filters spaced approximately 4 Hz apart beginning at the lowest frequencies increasing to approximately 24 Hz between adjacent filters at the highest frequencies. While this many filters may present a computational challenge to real time operation, it is noted that this is a relatively small number compared to the approximately 10-12,000 hair cells of the cochlea over the same frequency range.

Figure 4 is a graphic illustration of a portion of the filter arrangement around 100 Hz. It can be seen from Figure 4 that because of the filter spacing of about 4 Hz, the true frequency of any signal in this frequency range will be within about 2 Hz of the reported frequency. It is understood that better frequency resolution may be obtained by increasing the number of filters such as by using the frequency discrimination of higher loudness levels in Figure 3. It is also noted that satisfactory phone recognition might also be obtained with somewhat less frequency resolution; i.e., greater spacing between filters.

The output of each of the bandpass filters is computed for each sample. At an 8 KHz sampling rate, the spacing between samples is .125 ms. Modern digital signal processing chips arranged in parallel can provide the processing power required for real-time operation. For example, Loral Space Information Systems has developed an arrangement of five C-programmable Texas Instruments TMS320C30 DSP chips on two plug-in boards (marketed by California Scientific Software as the BrainMaker Professional Accelerator) that can provide adequate computing speed to solve several hundred filters in real

time. Alternatively, more compact integrated circuits can be specially designed for the purpose.

The maximum absolute amplitude of each frequency band is determined over a short time interval. The length of that interval is a balance between shortness for accuracy in representing the dynamics inherent in speech patterns, and length to accurately reflect the amplitude of low frequencies. The duration of one complete wave of a 1 KHz tone is 1 ms. One wave of a 500 Hz tone is 2 ms and that of a 250 Hz tone 4 ms. However, a half-wave of 125 Hz tone, the pitch of a typical male voice, is also 4 ms and will contain the maximum value attained in the full wave. In the preferred embodiment of the invention, a constant interval of 4 ms is employed over which to evaluate the maximum absolute value of the amplitude of each frequency band. A longer time period could be used but the presence of lower frequencies does not appear to contribute significantly to intelligibility. Likewise, shorter intervals could be used for higher frequencies, thereby achieving greater accuracy in the time domain for those frequencies. The additional complexity resulting may be tolerated for some speech analysis applications but was not considered cost-effective in the preferred embodiment.

The output of the spectral analysis filter arrangement is a representation of the speech signal leaving the vocal tract. However, it is well-known that human hearing does not have a flat frequency response. It is considerably less sensitive to the lowest frequencies in the speech spectrum than the higher ones. Figure 5 illustrates the relative sound level intensity required for perceived equal loudness. Referring to the "10 Loudness Level(phons)" curve of Figure 5, it can be seen that a signal of about 30db greater sound pressure level is required for a 100 Hz signal to produce the same perceived loudness as a 1000 Hz signal. The present invention modifies the output of the filter bank to compensate for the frequency response of the ear. In the

preferred embodiment, each of the outputs of the bandpass filters is multiplied in the spectrum analyzer 4 by the inverse of the "10 Loudness Level(phons)" curve of Figure 5. This increases the amplitude of the higher frequencies relative to the lower ones. It can be seen that this has the effect of somewhat compensating for the phenomenon of the amplitudes of the pitch harmonics declining at 6db per octave as discussed previously.

Neural Network Phone Identifier

The neural network phone identifier 5 receives the output of the spectrum analyzer 4 and inputs it into its main processor, an artificial neural network that has been trained to identify the speech sounds or phones which make up the speech stream. The artificial neural network is trained by a method described in detail below to recognize not only phonemes but all legitimate speech sounds in a language including such sounds as murmurs occurring before a nasal like "M" and "N", and allophones (or variants) of phonemes; e.g., as is well-known by those skilled in the art of phonetics the acoustic spectrum of the "Z" occurring at the beginning of a syllable is often different from that of a "Z" occurring before a silence. While it is generally accepted that there are only about 40 to 45 phonemes in American English, there are over a hundred different sounds in the language as just illustrated. The term phone is used herein to refer to all such legitimate speech sounds.

The present invention makes use of a fuzzy set concept of phones. In this concept, every sound made during speech has a probabilistic membership in all fuzzy phone sets. However, it is only when a particular sound's probability of being in a given set is sufficiently high and it's probability of being in any other set is sufficiently low is it labelled by the system as belonging to the given phone set. The differences between this concept and other concepts used in prior art is

illustrated in Figure 6A-6C. In Fig. 6A, all phones (or phonemes) in a speech stream are contiguous; i.e., where one phone (or phoneme) ends, the next one is considered to begin. Furthermore, all sounds in the stream are a part of some phone (or phoneme). In Fig. 6B, a sound can be part of a phone (or phoneme) or it can occur during a transition from one phone (or phoneme) to the next. However, the occurrence of a phone (phoneme) is a discrete event; the sound either is or is not a phone (or phoneme) -- the probability is either zero or one.

Fig. 6C illustrates that sounds in the speech stream can have a probabilistic membership in more than one phone fuzzy set. This follows from the fact that the vocal tract is a variable configuration mechanical device that is constantly being re-shaped to produce the desired sound. There is a unique target position of the vocal tract for each phone. During speech, sounds are continually being produced as the vocal tract is reconfigured to successive positions. Figures 7A-7E are illustrations of estimated articulatory positions during pronunciation of the word "caw". Fig. 7A is the estimated target position for the phoneme /K/ and Fig. 7E is the estimated target position for the phoneme /AO/ ("A Course in Phonetics"; Peter Ladefoged; Harcourt Brace Jovanovich College Publishers; Fort Worth, Texas; 1993). Figs. 7B-7D are some estimated positions of the vocal tract assumed during transition between the two target positions. It is clear that as the vocal tract shape is going away from the target position for the /K/, the sound produced will be less and less like that of the /K/. Likewise, as the shape approaches that of the /AO/, the sound produced will be more and more like that of the /AO/. In between the two target positions, the sounds will have varying similarities to the two target phonemes and indeed may have some similarities to other phones.

The artificial neural network is trained by a method described in detail below to identify when a phone is represented by the sound occurring in each 4 ms interval. It does this by solving the matrices representing the network weights applied to the spectral input and computing the probabilities that the sound represents each of the phones. If the probability for one of the phones exceeds a specified threshold, and the probabilities for all other phones are less than one minus that threshold, then the signal in that interval is identified as the phone exceeding the threshold. In one embodiment of the invention, the BrainMaker Professional neural network software produced by California Scientific Software is used both for the training and solution of networks. Other mechanisms for solving neural networks are available such as specialized neural chips with the result that alternative designs for implementing the invention in hardware are possible.

Artificial neural networks have been applied successfully to a variety of pattern recognition and correlation tasks. Methods of configuring, training and solving artificial neural networks are well-known to those skilled in the art. In order to apply one effectively to phone recognition, methods of providing it information necessary and sufficient for it to be able to recognize the speech sounds of an arbitrary speaker are required. Two conditions must be satisfied for accurate recognition. First, the description of the speech signal presented to the artificial neural network (for training and recognition) must be of sufficiently high resolution to allow it to distinguish between phones in a relatively crowded speech band. And second, the network must have previously been trained with the speech samples of a sufficient number and diversity of speakers of the language to ensure that the speech patterns on which it is trained are representative of the speech patterns of the full population. The spectrum analyzer 4, being designed

to provide resolution response and resolution similar to that of human hearing, satisfies the first condition. Regarding the second condition, the empirical results obtained in training the neural network phone identifier 5 for reducing this invention to practice show that speech samples from hundreds of speakers are required to achieve adequate coverage of male and female speakers with low to high pitch voices and a wide range of individual linguistic mechanisms. The numbers of different speakers required is discussed further below in connection with training the neural network. Figures 8A and 8B show the spectra of a few of the hundreds of examples of the vowels "AH" (as in "nut") and "OW" (as in "note") presented to the ANN for training. As can be seen from Figures 8A-8B, there is not only a great range of variation within a given phone but a great deal of similarity between the two phones.

Artificial neural networks typically have an input layer of neurons, an output layer and one or more hidden layers. A schematic diagram of the preferred embodiment of the phone recognition network is shown in Figure 9. The output layer of neurons is simply each of the phones of the spoken language. The input layer is spectral data for the current time interval and a previous one. As shown in Figure 9, the first neuron represents a measurement of the speech signal input level. The remaining neurons are two sets of input data which capture the rapidly changing dynamics of some phones such as stops by describing the signal spectrum at a previous interval and at the current one. The separation between the two intervals is selected to emphasize the differences in the spectra. In the preferred embodiment, the separation is 32 ms. It is understood that the optimal separation may be different for different languages and even for different dialects and regional accents in a given language. In each of the two sets, the first neuron gives the maximum amplitude of any frequency occurring in that

time interval and the remaining ones describe the signal spectrum relative to that maximum. As indicated previously, an artificial neural network may incorporate one or more hidden layers of neurons. Those skilled in the art of artificial neural network construction will recognize that no dependable theories or rules-of-thumb have yet been devised to determine either the optimum number of hidden layers or the optimum number of neurons in a hidden layer. In accordance with standard practice in the field, the number of neurons in the hidden layer(s) is determined empirically by testing hidden layers with different numbers of neurons to identify the one yielding the best performance in terms of accuracy in correctly identifying the phones in the speech signals of speakers not included in the population of those on which the network was trained.

Training the Neural Network

Training the neural network includes preparing data to represent the speech characteristics of as much of the expected user population as possible. Speech samples are recorded using sets of words to be uttered that contain in each set one or more examples of each of the specific phones desired. One way of training a system for the one hundred plus phones in American English is to train the neural network on individual sets of approximately ten phones each and combining those sets into larger and larger training sets. It is important to include speakers in each training set whose collective voices span the range of pitch frequencies of voices expected to be encountered in the application. For example, if only men's voices are needed, a range from about 60 to about 150 Hz should be adequate; if only women's voices are needed, a range of about 130 to 350 Hz will be required. If children's speech also is to be recognized, the range will be extended perhaps as high as 400 Hz.

It is important to have a more or less uniform distribution of numbers of pitch voices over the desired range. The preferred embodiment of the disclosed invention has approximately forty frequency bands over the range of voice pitches. It can be estimated statistically that about fifty different speakers for each voice pitch should yield high confidence of population representation. It will be observed in collecting speech samples for training the system that voice pitches will tend to cluster about certain frequencies in approximately normal distributions separately for men and women (and children also if included). In collecting speech samples for training the proof-of-principle system of the present invention, it was found for that particular sample population that there were fewer men's voices between 60-100 Hz and 130-150 Hz than between those ranges. Likewise there were fewer women's voices in the 150-180 Hz and 250-350 Hz ranges than in between. It can be expected to find a surplus of mid-frequency pitches to be discarded and additional effort required to get a sufficient number of high and low pitch voices to achieve desired uniformity in pitch distribution.

The most important part of the training process is to select the "best" times to represent each phone in a sample word; i.e., the times at which the probabilities are highest that the spectra belong to the fuzzy sets of the sample phones. Referring again to Figures 6A-6C, those times are the peaks of the curves for the 3-phone word shown in Fig. 6C. It is extremely helpful in selecting the times to view the output of the spectrum analyzer in graphical form. Figures 10A-10C are a high resolution spectrograms for the word "KNOW" uttered by subject JA9. (It can be observed from the figure that the subject is probably a woman since the voice pitch is about 180 Hz.) The duration of the displayed portion of the recording is 600 milliseconds; the figure is split into

three 200ms parts for convenience of display. Each tick mark at the top edge of each part represents 20 ms.

Both the murmur before, and the weak plosive release of, the phone "N" around 300 ms are clear. Thus the selection of the "best" times for these phones is facilitated. Selection of best times for other phones such as vowels may not be so clear cut. This subject, like many others from whom speech samples were taken, inserted the phoneme "AH" (as in "nut") between the "N" and the "OW" so that the pronounced word was N:AH:OW. Thus the phoneme "OW" does not occur at around 480 ms as might be supposed from Figure 10 (and if one is not aware as phoneticians are that the phoneme /AH/ is frequently inserted) but instead around 576 ms.

A representative output of the neural network phone identifier 5 for the sample word KNOW.JA9 is displayed in Figures 11A-11D. It can be seen from Figures 11A-11D that at some times (such as around 200 ms) the signal has a significant probability of belonging to more than one phone set as was discussed in connection with Figure 6C. Likewise note the increasing probability for the murmur (xN) before the N, then its probability decreasing while the probability of the N increases. Subsequently the probability of the N decreases while the probability of the AH increases, and then the probability of the AH decreases while the probability of the OW increases.

The times selected initially for the thousands of phone examples in a given training set perhaps will not be the ones representing the times of maximum probability for at least some of the phones. During training, the neural network is looking for consistent patterns. Therefore, after training, the trained neural network should be applied to the sample words and significant differences between the phone input times and those identified by the neural network as being the highest probability times spotted. The non-optimum sample times then can be adjusted and the training repeated. This process should

be iterated until the differences reach an acceptably low level. In addition, testing of the system on new subjects after the system is trained may result in low probabilities of phone recognition for some speakers. The data for such speakers can be fed back into further training of the system to improve performance.

This same technique is used when training the system for a new language. Speech samples from speakers of the new language are tested using the existing trained network in order to identify those phones for which the system already gives satisfactory results versus those that need to be trained specifically for that language. It is understood that those phones in the new language that are not common to a previous language will have to trained on speech samples from the new language.

Phoneme Integrator

The artificial neural network identifies which phone (if any) occurs in each time interval. However, some phonemes such as vowels are of sustained duration. One function of the phoneme integrator 6 is to separate legitimate phones from non-phonetic transitions by imposing a requirement for a pre-determined minimum number of consecutive identifications to confirm recognition. In the preferred embodiment of the disclosed invention, eight consecutive identifications (equivalent to 32 milliseconds duration) is required to confirm recognition of a vowel, three consecutive identifications for semi-vowels and fricatives and only one for stops and other plosives. Another of its functions is to ensure that both a murmur phone (of sufficient duration) and a release phone are present for phonemes such as voiced stops before recognition is confirmed. The output of the phoneme integrator is the phonemic representation of the speech stream.

Phoneme String Translator

The function of the phoneme string translator 7 is to identify, separate and display (or output to a file) the spoken language words represented by the phoneme string. The major components of the translator are a phonemic-spoken language dictionary and a computer program that uses that dictionary to convert the phoneme string into words spelled in the spoken language. An important feature of the dictionary is the use of multiple phonemic entries for many of the natural words. This is rendered necessary because (a) people with different accents often pronounce a given word differently and (b) transitions from one phone to another are sometimes a third phone. An example of (a) is the often different pronunciation of the word "harbor" by natives of the Northeastern United States compared to those in the Midwest. An example of (b) is the frequent transitional "AH" between an "N" and an "OW" and the insertion of a "W" between an "OW" and an "AH" so that the word "Noah" can have at least the phonemic spellings of /N:OW:AH/, /N:AH:OW:AH/, /N:OW:W:AH/ AND /N:AH:OW:W:AH/. The phonemic-spoken language dictionary has, and uses, all these entries to separate the phoneme string into spoken language words.

The computer program design is based on identifying words in the context of a longer string of phonemes and to specifically address and account for co-articulation effects such as gemination. Before describing the program it is useful first to identify a frequently occurring phonetic situation not addressed in the prior art. When one spoken word ends in a given phoneme, especially a stop or a fricative, and the next word begins with the same phoneme, the two phonemes are seldom enunciated separately. Identifying the location of word separation is made more complex for a speech recognition system than when such a situation does not obtain. For example, the utterance "bad dog" can not be properly separated without factoring in gemination of the ending and beginning "d".

Otherwise the alternatives are "bad og" and "ba dog"; neither of which identify both words correctly. In a small vocabulary application, such a situation may be avoided by restricting words included in the lexicon but can not be in the unlimited vocabulary application for which this invention is intended. It is noted that there are numerous phonemes that are gemination candidates including all of the stops and fricatives and some of the affricates.

The computer program is designed to anticipate possible gemination occurrences. A logical diagram of the computer program is shown in Figure 12. The approach involves using a phoneme string longer than any single word likely to be encountered. The preferred embodiment of the invention is based on a 20 phoneme string length (called MaxString in procedure 10 of Figure 12). The first 20 phonemes in an utterance (or the actual length if the utterance is less than 20 phonemes long) is examined in procedure 11 to find the longest possible first word. If that word does not end in a gemination candidate, it is output in procedure 16, the next phoneme becomes the new starting point in procedure 17, the 20 phoneme length is replenished in procedure 10, and the process repeated. If the longest first word does end in a gemination candidate, procedure 13 extends temporarily the MaxString by a number of phonemes equal to the number of phonemes in the test word, then procedure 14 determines whether there is a following word in the extended MaxString. This indicates that the phoneme following the gemination candidate was not co-articulated with the last phoneme in the preceding word. If there is a following word, procedure 16 outputs the test word, the next phoneme becomes the new starting point in procedure 17, the 20 phoneme length is replenished in procedure 10, and the process repeated. If there is not a second word commencing after the test word (indicating that co-articulation has occurred), procedure 15 inserts a duplicate of the co-articulation candidate

phoneme at that point. As before, procedure 16 outputs the test word, the next phoneme becomes the new starting point in procedure 17, the 20 phoneme length is replenished in procedure 10, and the process repeated. This set of procedures is repeated as long as there are phonemes produced by the phoneme integrator 6.

It should be noted that although the basic design of the system described above assumes that the user normally will select a specific language to be transcribed prior to use, the system can be modified to automatically determine which of the languages within its repertoire is being spoken and to select the appropriate artificial neural network and language dictionary for use. This can be accomplished by processing a brief initial portion of the speech, say 5 to 10 seconds in duration, through each of the languages to identify the language that produces a string of real words. The language for which the system identifies a string of real words is selected and the system operates from that point on as described above.

System Implementation in Hardware

The method and system disclosed herein may require concurrent processing for real time operation unless implemented on a "super computer"; however, it is intended primarily for widespread use and the preferred implementation is on a "personal computer" or "workstation" class of machine. While the equipments of several manufacturers may have suitable characteristics for some of the various components of the system, a particular arrangement as shown in Figure 13A and 13B will be described for purposes of illustration. As mentioned above, Loral Space Information Systems has developed an arrangement of five C-programmable Texas Instruments TMS320C30 DSP chips on two plug-in boards 105 and 106 that can provide adequate computing speed for solving the equations for several hundred narrow bandpass filters in real time. A second set of boards 103 and 104 can be

dedicated to solving the neural network equations. These two sets of boards can be installed for example in a Compaq SystemPro Model 66M microcomputer which has provision for two independent processor boards 110 and 111 that can share the same memory installed on boards 108 and 109. One of these processors accomplishes the phoneme integration 6 function while the other serves as both as the control processor for language selection and to provide Phonemic-to-Spoken Language Translation and text output. Another plug-in board 107 such as the Media Vision Pro Audio Spectrum 16 can provide the analog-to-digital conversion function and its accompanying software can support waveform display and editing for assembling speech samples for language training and testing. The SystemPro computer has two remaining empty slots available.

Claims

1. A multi-language speech recognition system comprising

an analog-to-digital converter for converting speech sounds into digital information,

an analyzer for receiving said digital information and determining, with the frequency discrimination and frequency response of human hearing, a spectrum of said speech sounds,

a phone identifier for receiving said spectrum from said spectrum analyzer, said phone identifier comprising a network for identifying which phones, if any, occur within a specified time interval of said spectrum, said network being capable of recognizing phones of a predetermined language,

a phoneme integrator for separating phones from non-phonetic transitions identified by said phone identifier by detecting a predetermined minimum number of consecutive identifications of said identified phones to confirm recognition, said phoneme integrator providing as an output a phoneme string representative of identified phones from said spectrum, and

a phoneme string translator for identifying, separating, and displaying or storing to a file, the human language words represented by said phoneme string, said phoneme string translator comprising a phonemic-spoken language dictionary and a program that uses said dictionary to convert the phoneme string into text of the spoken language.

2. The system of Claim 1, wherein said network is pretrained to recognize phones in any one of several given human languages.

3. The system of Claim 1, wherein said system accommodates multiple pronunciations of a word.

4. The system of Claim 1, wherein said system transcribes continuous conversational speech of an arbitrary speaker into one of several languages.

5. The system of Claim 1, wherein said network identifies a time frame within said spectrum which most closely approaches a vocal tract configuration of a target position of a given phone.

6. The system of Claim 1, wherein said system addresses and accounts for coarticulation effects such as gemination which occur in conversational speech.

7. A method for multi-language speech recognition comprising the following steps

receiving an analog speech sound input and converting said input into a digital output,

receiving said digital output and determining, with the frequency discrimination and frequency response of human hearing, a spectrum of said speech sounds,

receiving said spectrum from said spectrum analyzer, and identifying which phones, if any, occur within a specified time interval of said spectrum, by comparing said spectrum with information in a network to recognize phones of a predetermined language,

separating phones from non-phonetic transitions identified by said comparison by detecting a predetermined minimum number of consecutive identifications of said identified phones to confirm recognition, and providing as an output a phoneme string representative of identified phones from said radiated spectrum, and

identifying, separating, and displaying or storing to a file, the human language words represented by said phoneme string, by use of a phoneme string translator comprising a phonemic-spoken language dictionary and a program that uses said dictionary to convert the phoneme string into text of the spoken language.

8. The method of Claim 7, wherein said network is pretrained to recognize phones in any one of several given human languages.

9. The method of Claim 7, wherein said method makes efficient use of common phones which exist in various human languages when adding additional language capabilities to said method.

10. The method of Claim 7, wherein said method accommodates multiple pronunciations of a word.

11. The method of Claim 7, wherein said method transcribes continuous conversational speech of an arbitrary speaker into one of several languages.

12. The method of Claim 7, wherein said network identifies a time frame within said spectrum which most closely approaches a vocal tract configuration of a target position of a given phone.

13. The method of Claim 7, wherein said method addresses and accounts for coarticulation effects such as gemination which occur in conversational speech.

14. A multi-language speech recognition system comprising

means for receiving audio speech signals in a predetermined language and for converting them into corresponding electrical signals,

an analog-to-digital converter for sampling said frequencies at a rate at least twice as high as a predetermined maximum frequency of interest in said signals,

a spectrum analyzer for accepting sets of samples from said analog-to-digital converter over a time interval of between 1 millisecond to 8 milliseconds, and for

providing an analysis of the spectral content of each said set of samples simulating the frequency discrimination and sensitivity response characteristics of human hearing,

an artificial neural network for identifying whether each said set of samples probably represents the audio spectrum of one of a predetermined set of phones belonging to said spoken language,

an integrator for integrating sufficient predetermined minimum numbers of said probabilistic identifications of successive said sets of samples to confirm existence and recognition of said phones,

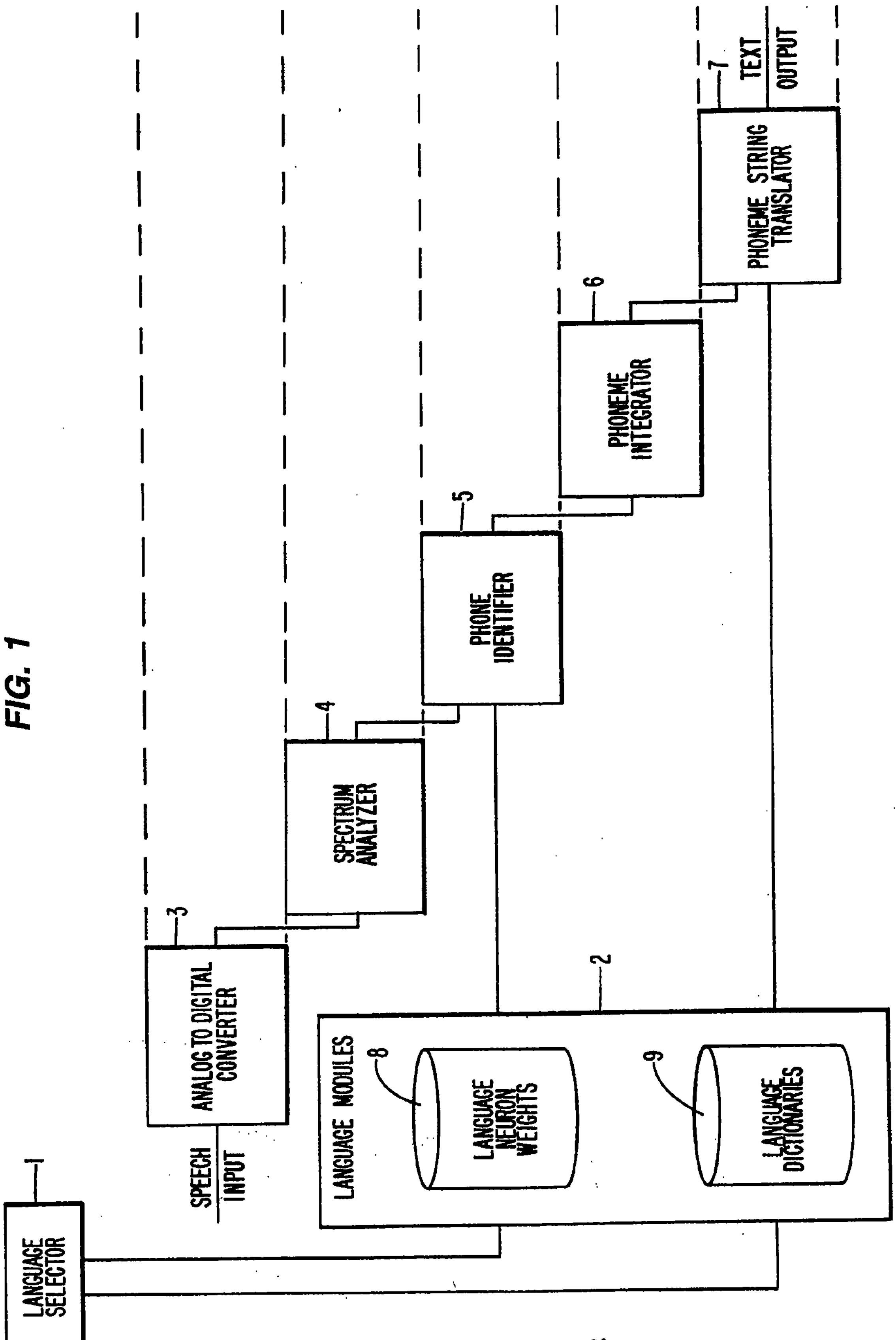
means for integrating said phones into phonemes in said spoken language,

a translator for translating sequences of said phonemes into words of said spoken language,

said translator including means for (1) interpreting transitions between two legitimate speech sounds that are third legitimate speech sounds and (2) identifying an unpronounced speech sound when said speech sound is co-articulated with a neighboring speech sound, and

means for displaying, printing and/or storing text corresponding to the translated words.

FIG. 1


LANGUAGE SELECTOR
1
SPEECH INPUT
ANALOG TO DIGITAL CONVERTER
3
SPECTRUM ANALYZER
4
PHONE IDENTIFIER
5
PHONEME INTEGRATOR
6
PHONEME STRING TRANSLATOR
7
TEXT OUTPUT
LANGUAGE MODULES
2
8
LANGUAGE NEURON WEIGHTS
9
LANGUAGE DICTIONARIES

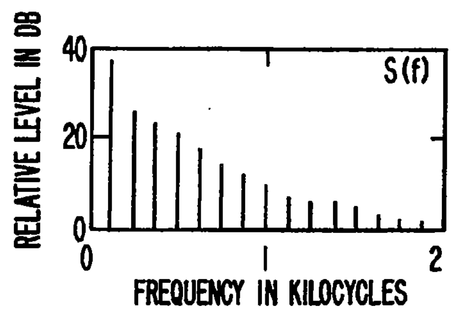


FIG. 2A

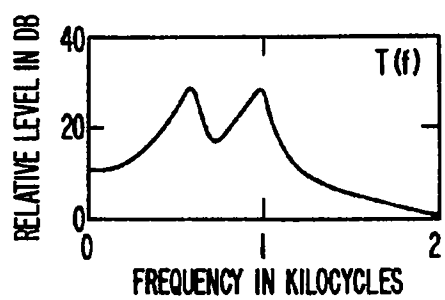


FIG. 2B

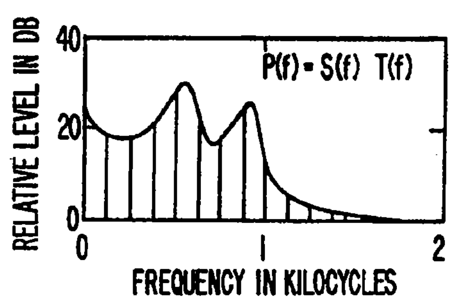


FIG. 2C

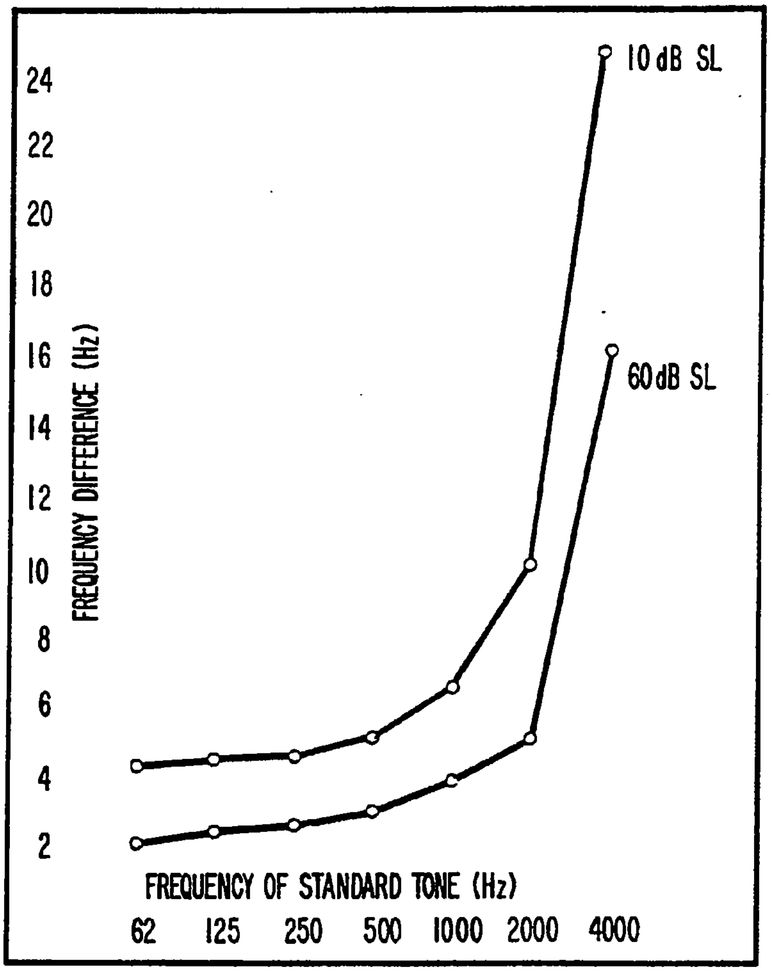

10 dB SL
60 dB SL
FREQUENCY DIFFERENCE (Hz)
24
22
20
18
16
14
12
10
8
6
4
2
FREQUENCY OF STANDARD TONE (Hz)
62
125
250
500
1000
2000
4000

FIG. 3

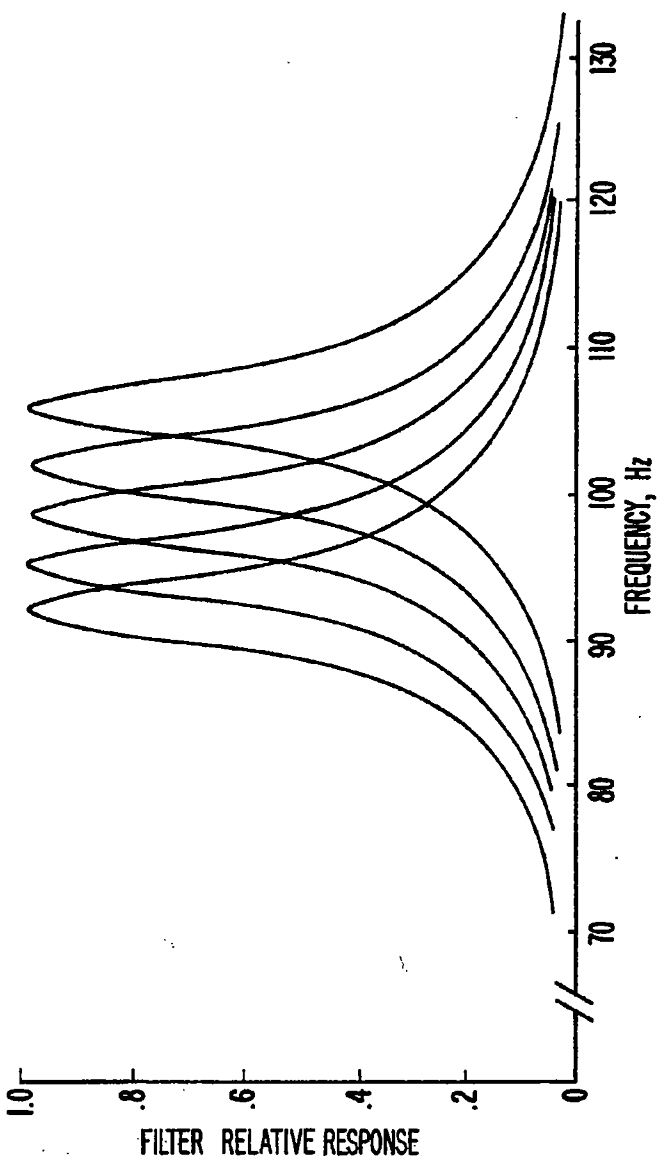
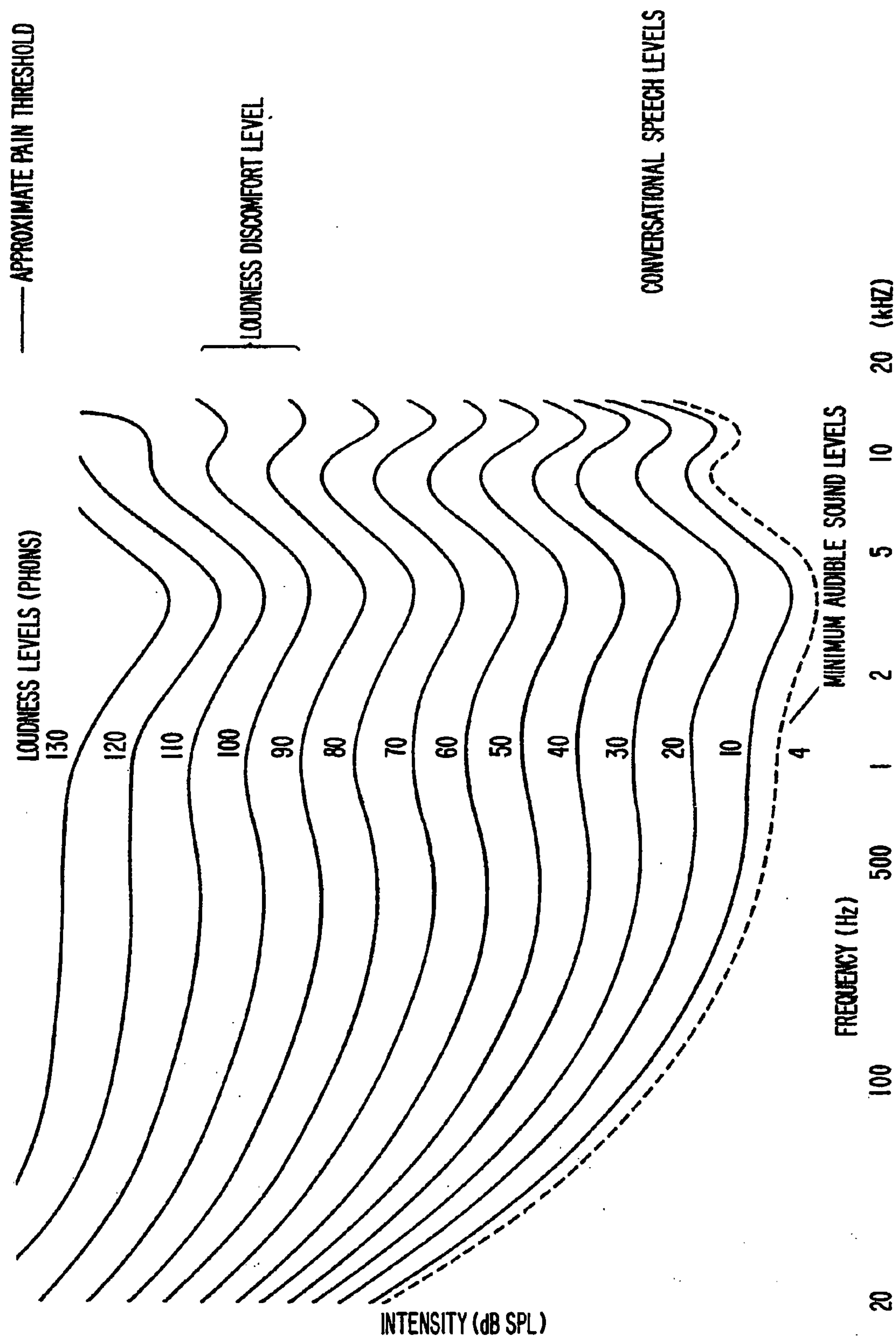

FILTER RELATIVE RESPONSE
1.0
.8
.6
.4
.2
0
70
80
90
100
110
120
130
FREQUENCY, Hz

FIG. 4

FIG. 5


APPROXIMATE PAIN THRESHOLD
LOUDNESS DISCOMFORT LEVEL
CONVERSATIONAL SPEECH LEVELS
LOUDNESS LEVELS (PHONS)
130
120
110
100
90
80
70
60
50
40
30
20
10
4
MINIMUM AUDIBLE SOUND LEVELS
INTENSITY (dB SPL)
140
120
100
80
60
40
20
0
20
100
500
1
2
5
10
20 (kHZ)
FREQUENCY (Hz)

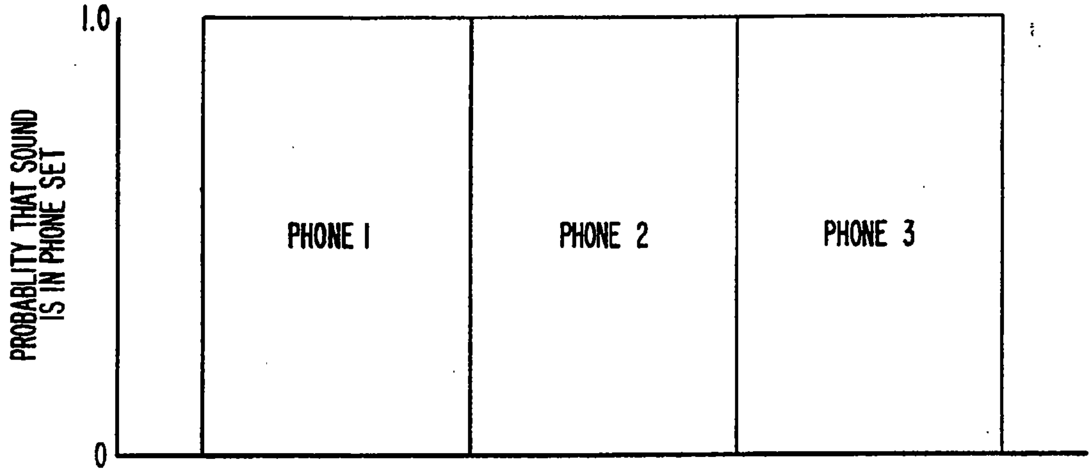


FIG. 6A

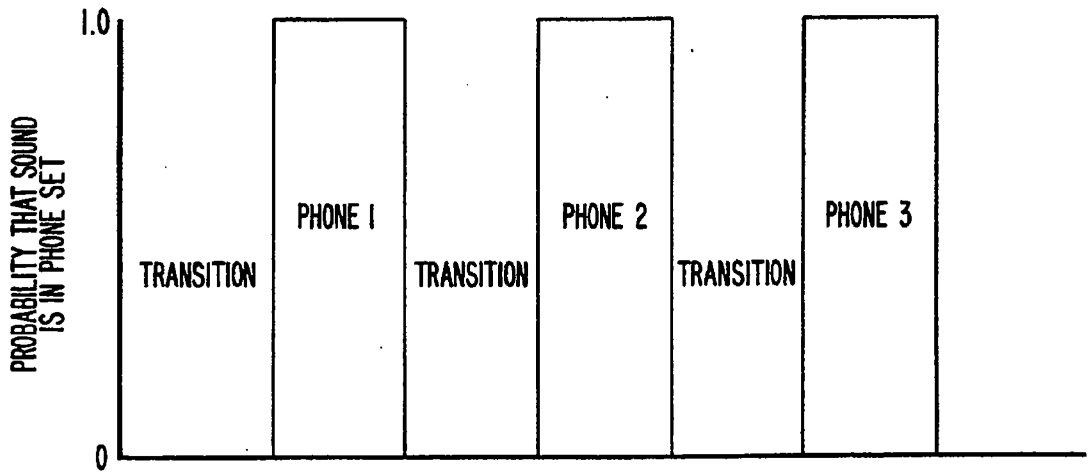


FIG. 6B

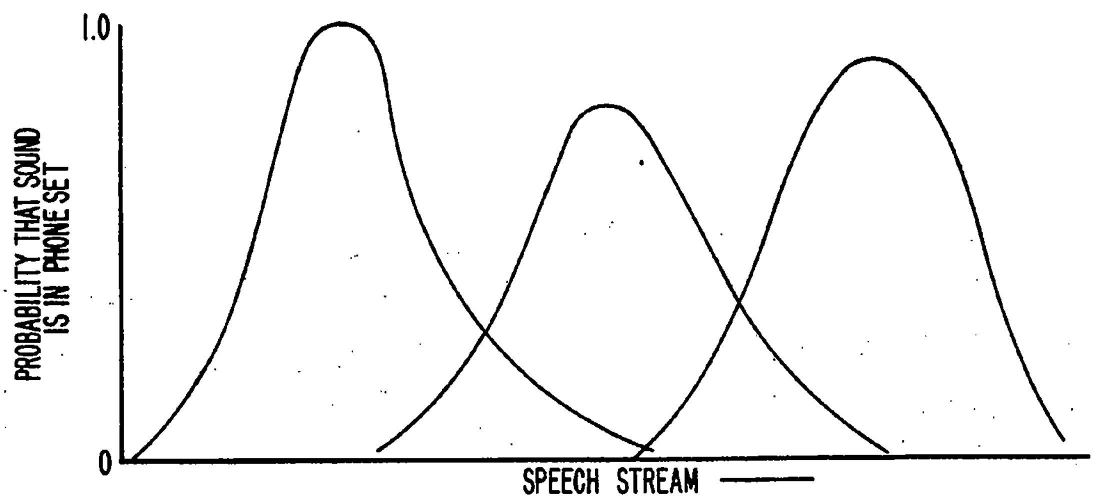


FIG. 6C

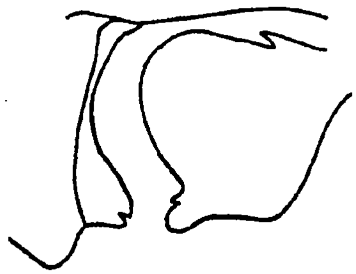


FIG. 7E

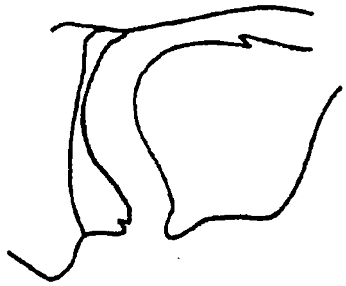


FIG. 7D

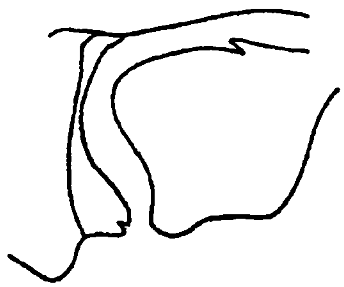


FIG. 7C

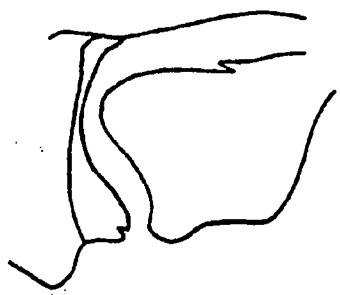


FIG. 7B

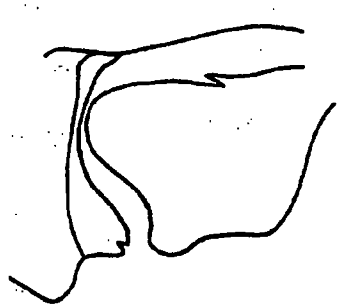


FIG. 7A

FIG. 8A

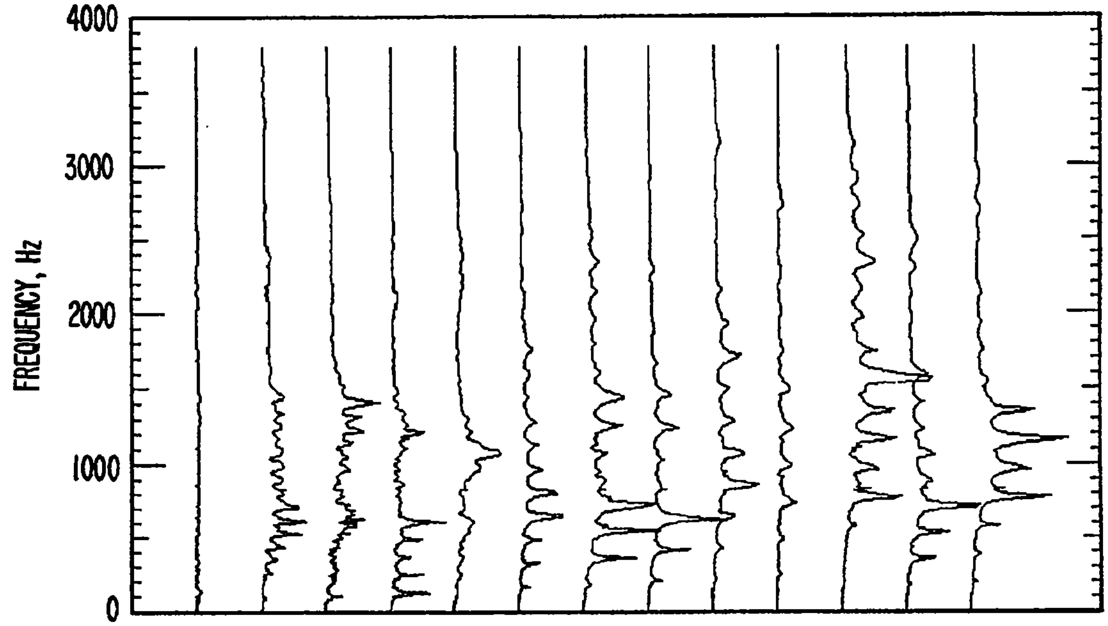
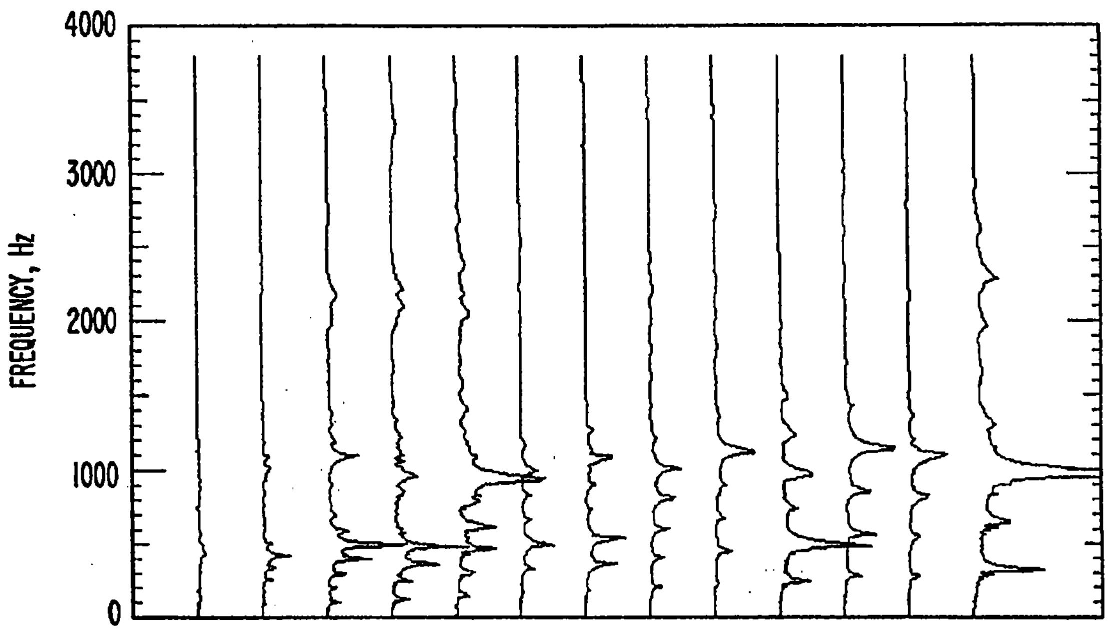

4000
3000
2000
1000
0
FREQUENCY, Hz

FIG. 8B


4000
3000
2000
1000
0
FREQUENCY, Hz

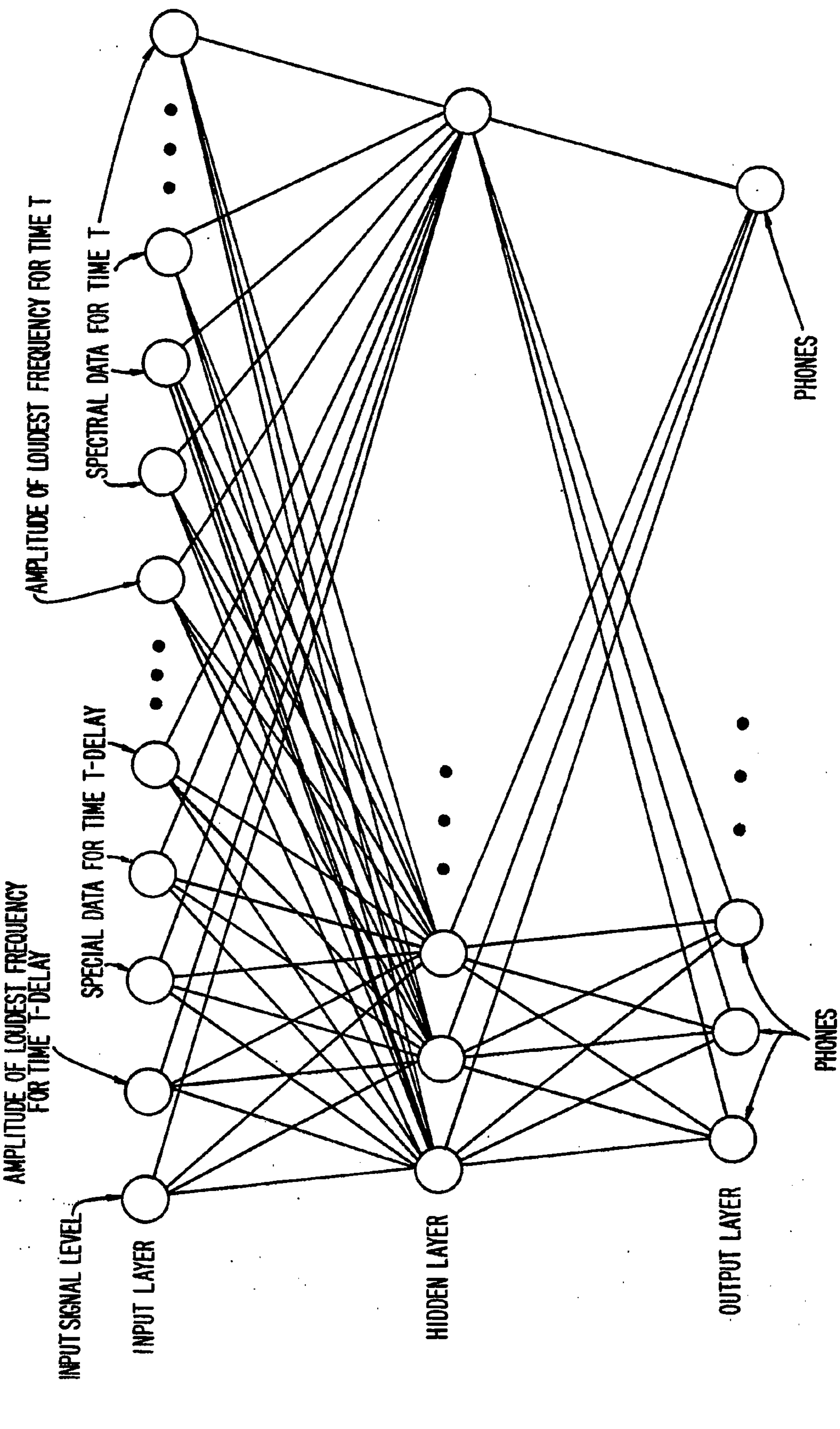

AMPLITUDE OF LOUDEST FREQUENCY FOR TIME T
SPECTRAL DATA FOR TIME T
SPECIAL DATA FOR TIME T-DELAY
AMPLITUDE OF LOUDEST FREQUENCY FOR TIME T-DELAY
INPUT SIGNAL LEVEL
INPUT LAYER
HIDDEN LAYER
OUTPUT LAYER
PHONES
PHONES

FIG. 9

FIG. 10A

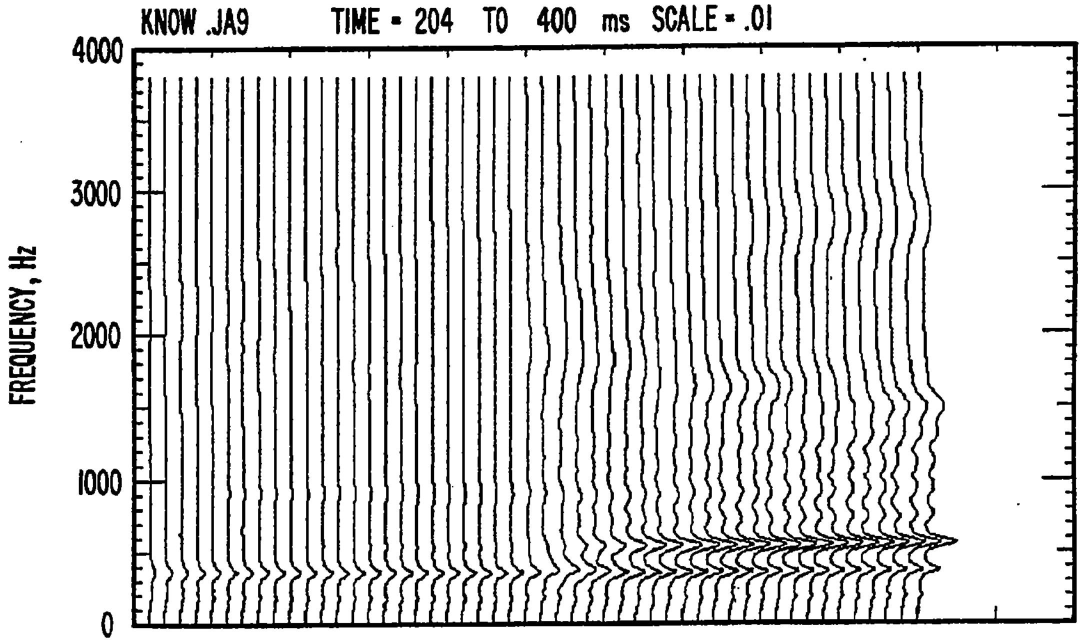


FIG. 10B

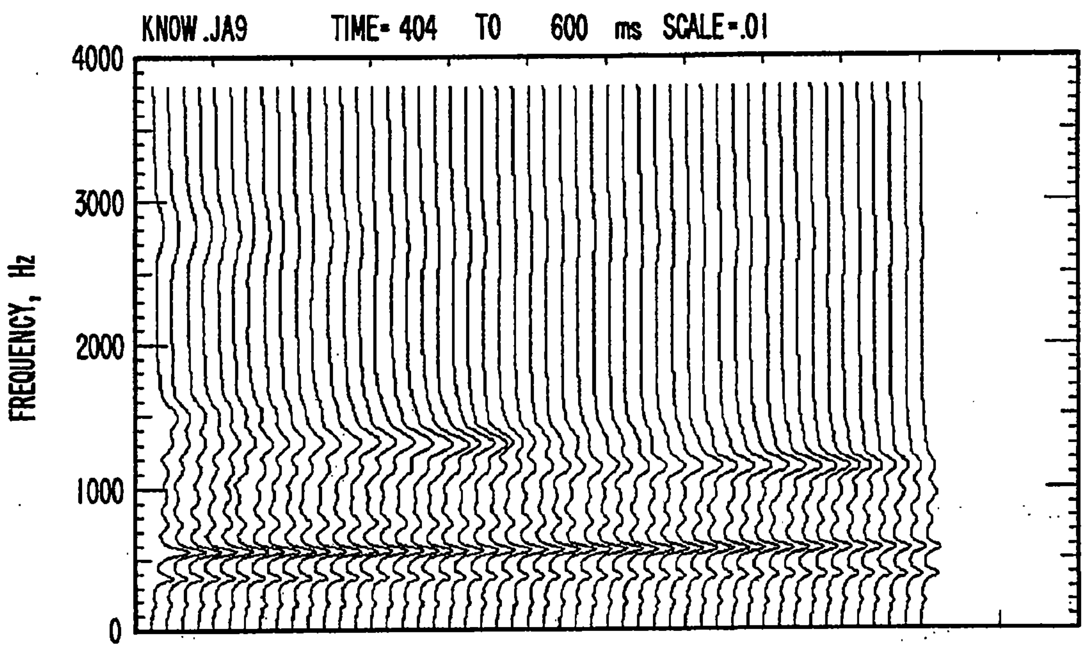
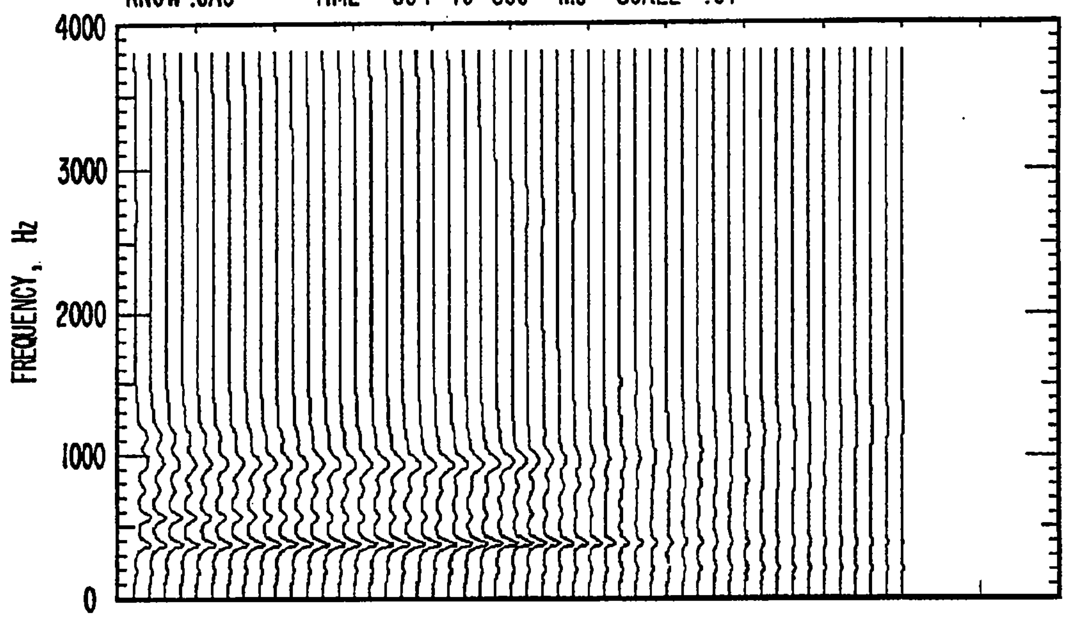


FIG. 10C


KNOW.JA9 TIME = 604 TO 800 ms SCALE = .01
FREQUENCY, Hz
4000
3000
2000
1000
0

4 ..	0.013AH	0.000IY	0.000OW	0.997N	0.000R	0.009W	0.000xN	0.000Zr	0.275Zc
8 ..	0.099AH	0.000IY	0.000OW	0.970N	0.000R	0.006W	0.000xN	0.000Zr	0.549Zc
12 ..	0.073AH	0.000IY	0.000OW	0.827N	0.000R	0.001W	0.000xN	0.000Zr	0.991Zc
16 ..	0.276AH	0.000IY	0.000OW	0.522N	0.000R	0.000W	0.000xN	0.000Zr	0.988Zc
20 ..	0.297AH	0.000IY	0.000OW	0.418N	0.000R	0.000W	0.000xN	0.000Zr	0.998Zc
24 ..	0.308AH	0.000IY	0.000OW	0.710N	0.000R	0.000W	0.000xN	0.000Zr	0.9987c
28 ..	0.105AH	0.000IY	0.000OW	0.779N	0.000R	0.000W	0.000xN	0.000Zr	1.000Zc
32 ..	0.141AH	0.040IY	0.008OW	0.000N	0.000R	0.000W	0.006xN	0.114Zr	0.247Zc
36 ..	0.159AH	0.010IY	0.028OW	0.001N	0.000R	0.000W	0.042xN	0.621Zr	0.093Zc
40 ..	0.153AH	0.007IY	0.043OW	0.001N	0.000R	0.006W	0.063xN	0.416Zr	0.416Zc
44 ..	0.234AH	0.002IY	0.019OW	0.001N	0.000R	0.030W	0.049xN	0.231Zr	0.833Zc
48 ..	0.252AH	0.004IY	0.023OW	0.001N	0.000R	0.029W	0.016xN	0.151Zr	0.917Zc
52 ..	0.382AH	0.008IY	0.019OW	0.000N	0.000R	0.009W	0.009xN	0.089Zr	0.951Zc
56 ..	0.294AH	0.008IY	0.016OW	0.001N	0.000R	0.010W	0.015xN	0.040Zr	0.940Zc
60 ..	0.194AH	0.003IY	0.008OW	0.000N	0.000R	0.008W	0.001xN	0.023Zr	0.988Zc
64 ..	0.146AH	0.003IY	0.007OW	0.001N	0.000R	0.004W	0.002xN	0.032Zr	0.952Zc
68 ..	0.240AH	0.002IY	0.021OW	0.001N	0.000R	0.006W	0.008xN	0.094Zr	0.948Zc
72 ..	0.280AH	0.001IY	0.035OW	0.003N	0.002R	0.010W	0.014xN	0.654Zr	0.361Zc
76 ..	0.203AH	0.002IY	0.046OW	0.002N	0.003R	0.008W	0.013xN	0.543Zr	0.503Zc
80 ..	0.293AH	0.003IY	0.049OW	0.002N	0.000R	0.034W	0.028xN	0.428Zr	0.732Zc
84Zc	0.246AH	0.002IY	0.062OW	0.003N	0.002R	0.014W	0.045xN	0.218Zr	0.808Zc
88 ..	0.194AH	0.003IY	0.040OW	0.001N	0.000R	0.005W	0.034xN	0.325Zr	0.722Zc
92 ..	0.169AH	0.001IY	0.042OW	0.002N	0.000R	0.006W	0.068xN	0.340Zr	0.818Zc
96 ..	0.096AH	0.001IY	0.028OW	0.002N	0.001R	0.025W	0.019xN	0.334Zr	0.952Zc
100..	0.222AH	0.002IY	0.031OW	0.001N	0.003R	0.021W	0.007xN	0.093Zr	0.973Zc
104..	0.136AH	0.002IY	0.024OW	0.002N	0.000R	0.038W	0.003xN	0.123Zr	0.973Zc
108..	0.190AH	0.002IY	0.023OW	0.010N	0.001R	0.022W	0.017xN	0.234Zr	0.877Zc
112..	0.158AH	0.001IY	0.013OW	0.026N	0.006R	0.008W	0.012xN	0.185Zr	0.904Zc
116..	0.106AH	0.002IY	0.024OW	0.006N	0.004R	0.026W	0.004xN	0.249Zr	0.947Zc
120..	0.041AH	0.003IY	0.006OW	0.002N	0.000R	0.014W	0.000xN	0.034Zr	0.995Zc
124..	0.031AH	0.005IY	0.007OW	0.001N	0.000R	0.015W	0.000xN	0.024Zr	0.996Zc
128..	0.022AH	0.004IY	0.004OW	0.001N	0.000R	0.004W	0.000xN	0.013Zr	0.997Zc
132..	0.058AH	0.006IY	0.017OW	0.001N	0.000R	0.025W	0.002xN	0.292Zr	0.951Zc
136..	0.042AH	0.004IY	0.005OW	0.004N	0.000R	0.013W	0.004xN	0.196Zr	0.912Zc
140..	0.045AH	0.000IY	0.002OW	0.011N	0.000R	0.011W	0.000xN	0.047Zr	0.972Zc
144..	0.038AH	0.000IY	0.000OW	0.012N	0.000R	0.002W	0.000xN	0.001Zr	0.997Zc
148..	0.040AH	0.000IY	0.000OW	0.007N	0.000R	0.003W	0.000xN	0.003Zr	0.986Zc
152..	0.211AH	0.000IY	0.001OW	0.012N	0.000R	0.002W	0.000xN	0.012Zr	0.931Zc
156..	0.101AH	0.000IY	0.001OW	0.002N	0.000R	0.003W	0.000xN	0.004Zr	0.995Zc
160Zc	0.164AH	0.000IY	0.002OW	0.009N	0.000R	0.009W	0.000xN	0.016Zr	0.967Zc
164 ..	0.202AH	0.002IY	0.013OW	0.001N	0.000R	0.043W	0.004xN	0.041Zr	0.983Zc
168 ..	0.474AH	0.004IY	0.029OW	0.001N	0.000R	0.024W	0.020xN	0.143Zr	0.870Zc
172 ..	0.136AH	0.003IY	0.030OW	0.001N	0.000R	0.164W	0.007xN	0.429Zr	0.849Zc
176 ..	0.144AH	0.003IY	0.015OW	0.002N	0.000R	0.060W	0.030xN	0.638Zr	0.424Zc
180Zc	0.101AH	0.004IY	0.016OW	0.001N	0.000R	0.050W	0.005xN	0.246Zr	0.745Zc
184 ..	0.180AH	0.006IY	0.047OW	0.001N	0.000R	0.020W	0.012xN	0.649Zr	0.431Zc
188 ..	0.115AH	0.019IY	0.059OW	0.004N	0.001R	0.005W	0.118xN	0.801Zr	0.040Zc
192Zc	0.018AH	0.037IY	0.036OW	0.034N	0.019R	0.258W	0.244xN	0.928Zr	0.013Zc
196 ..	0.026AH	0.075IY	0.079OW	0.045N	0.029R	0.743W	0.208xN	0.885Zr	0.029Zc
200 ..	0.011AH	0.055IY	0.084OW	0.019N	0.022R	0.871W	0.061xN	0.761Zr	0.005Zc
204 ..	0.011AH	0.069IY	0.219OW	0.009N	0.006R	0.544W	0.013xN	0.878Zr	0.002Zc
208 ..	0.002AH	0.006IY	0.076OW	0.005N	0.045R	0.477W	0.006xN	0.899Zr	0.003Zc
212 ..	0.000AH	0.001IY	0.033OW	0.003N	0.171R	0.441W	0.003xN	0.795Zr	0.002Zc
216 ..	0.001AH	0.000IY	0.027OW	0.002N	0.120R	0.742W	0.002xN	0.245Zr	0.001Zc
220 ..	0.002AH	0.000IY	0.011OW	0.004N	0.207R	0.407W	0.109xN	0.005Zr	0.001Zc

FIG. 11a

224 ..	0.000AH	0.001IY	0.002OW	0.005N	0.925R	0.011W	0.089xN	0.008Zr	0.000Zc
228 ..	0.000AH	0.009IY	0.002OW	0.006N	0.940R	0.011W	0.126xN	0.003Zr	0.001Zc
232 ..	0.000AH	0.013IY	0.002OW	0.020N	0.819R	0.029W	0.334xN	0.011Zr	0.004Zc
236 ..	0.001AH	0.016IY	0.006OW	0.046N	0.519R	0.062W	0.680xN	0.027Zr	0.011Zc
240 ..	0.001AH	0.018IY	0.006OW	0.070N	0.425R	0.037W	0.835xN	0.029Zr	0.008Zc
244 ..	0.001AH	0.015IY	0.010OW	0.052N	0.409R	0.046W	0.872xN	0.030Zr	0.016Zc
248xN	0.003AH	0.009IY	0.015OW	0.063N	0.242R	0.061W	0.906xN	0.027Zr	0.014Zc
252xN	0.005AH	0.011IY	0.019OW	0.066N	0.223R	0.054W	0.915xN	0.042Zr	0.026Zc
256xN	0.003AH	0.009IY	0.020OW	0.079N	0.204R	0.029W	0.952xN	0.028Zr	0.018Zc
260xN	0.004AH	0.010IY	0.026OW	0.117N	0.102R	0.061W	0.952xN	0.051Zr	0.020Zc
264xN	0.005AH	0.007IY	0.029OW	0.104N	0.102R	0.056W	0.935xN	0.066Zr	0.019Zc
268xN	0.004AH	0.007IY	0.034OW	0.098N	0.115R	0.051W	0.882xN	0.079Zr	0.023Zc
272xN	0.003AH	0.006IY	0.033OW	0.101N	0.113R	0.053W	0.906xN	0.064Zr	0.023Zc
276xN	0.004AH	0.006IY	0.035OW	0.125N	0.081R	0.053W	0.917xN	0.094Zr	0.019Zc
280xN	0.004AH	0.004IY	0.040OW	0.103N	0.050R	0.046W	0.960xN	0.068Zr	0.014Zc
284xN	0.003AH	0.005IY	0.032OW	0.147N	0.067R	0.035W	0.920xN	0.088Zr	0.020Zc
288xN	0.004AH	0.008IY	0.048OW	0.096N	0.034R	0.052W	0.964xN	0.066Zr	0.016Zc
292xN	0.005AH	0.007IY	0.045OW	0.093N	0.048R	0.040W	0.957xN	0.075Zr	0.019Zc
296xN	0.005AH	0.006IY	0.032OW	0.143N	0.044R	0.038W	0.951xN	0.089Zr	0.019Zc
300N	0.005AH	0.000IY	0.002OW	0.843N	0.035R	0.018W	0.241xN	0.125Zr	0.017Zc
304N	0.003AH	0.001IY	0.002OW	0.833N	0.016R	0.031W	0.187xN	0.216Zr	0.005Zc
308N	0.008AH	0.000IY	0.000OW	0.979N	0.001R	0.023W	0.009xN	0.072Zr	0.004Zc
312N	0.012AH	0.000IY	0.000OW	0.980N	0.002R	0.018W	0.003xN	0.092Zr	0.006Zc
316N	0.014AH	0.000IY	0.000OW	0.988N	0.004R	0.011W	0.006xN	0.064Zr	0.006Zc
320N	0.014AH	0.000IY	0.000OW	0.986N	0.007R	0.007W	0.009xN	0.077Zr	0.007Zc
324N	0.019AH	0.000IY	0.000OW	0.988N	0.003R	0.007W	0.008xN	0.092Zr	0.007Zc
328..	0.088AH	0.008IY	0.013OW	0.345N	0.060R	0.040W	0.183xN	0.563Zr	0.025Zc
332..	0.060AH	0.149IY	0.048OW	0.187N	0.186R	0.015W	0.189xN	0.425Zr	0.030Zc
336..	0.068AH	0.181IY	0.058OW	0.088N	0.181R	0.010W	0.091xN	0.453Zr	0.020Zc
340..	0.124AH	0.108IY	0.079OW	0.032N	0.203R	0.006W	0.037xN	0.404Zr	0.036Zc
344..	0.197AH	0.046IY	0.099OW	0.028N	0.067R	0.008W	0.056xN	0.585Zr	0.038Zc
348..	0.212AH	0.023IY	0.047OW	0.024N	0.043R	0.005W	0.045xN	0.443Zr	0.025Zc
352..	0.356AH	0.011IY	0.044OW	0.018N	0.020R	0.007W	0.038xN	0.512Zr	0.030Zc
356..	0.493AH	0.006IY	0.037OW	0.020N	0.016R	0.006W	0.034xN	0.442Zr	0.026Zc
360AH	0.629AH	0.006IY	0.027OW	0.028N	0.009R	0.005W	0.025xN	0.376Zr	0.029Zc
364AH	0.640AH	0.003IY	0.027OW	0.022N	0.014R	0.008W	0.015xN	0.375Zr	0.021Zc
368AH	0.722AH	0.002IY	0.024OW	0.015N	0.015R	0.007W	0.010xN	0.390Zr	0.023Zc
372AH	0.830AH	0.001IY	0.021OW	0.016N	0.023R	0.005W	0.005xN	0.289Zr	0.027Zc
376AH	0.841AH	0.001IY	0.017OW	0.021N	0.011R	0.007W	0.003xN	0.341Zr	0.048Zc
380AH	0.859AH	0.001IY	0.017OW	0.014N	0.025R	0.006W	0.003xN	0.290Zr	0.024Zc
384AH	0.910AH	0.000IY	0.009OW	0.015N	0.020R	0.005W	0.001xN	0.221Zr	0.033Zc
388AH	0.944AH	0.000IY	0.005OW	0.024N	0.013R	0.003W	0.001xN	0.158Zr	0.041Zc
392AH	0.948AH	0.000IY	0.005OW	0.028N	0.035R	0.001W	0.001xN	0.162Zr	0.033Zc
396AH	0.962AH	0.000IY	0.003OW	0.044N	0.014R	0.001W	0.001xN	0.104Zr	0.027Zc
400AH	0.966AH	0.000IY	0.004OW	0.022N	0.014R	0.001W	0.001xN	0.125Zr	0.025Zc
404AH	0.968AH	0.000IY	0.004OW	0.022N	0.006R	0.001W	0.001xN	0.153Zr	0.050Zc
408AH	0.957AH	0.000IY	0.007OW	0.015N	0.020R	0.001W	0.002xN	0.178Zr	0.037Zc
412AH	0.953AH	0.000IY	0.006OW	0.011N	0.014R	0.001W	0.002xN	0.172Zr	0.036Zc
416AH	0.958AH	0.000IY	0.008OW	0.005N	0.009R	0.001W	0.003xN	0.227Zr	0.035Zc
420AH	0.955AH	0.000IY	0.007OW	0.005N	0.002R	0.002W	0.003xN	0.282Zr	0.047Zc
424AH	0.951AH	0.000IY	0.010OW	0.004N	0.006R	0.000W	0.004xN	0.276Zr	0.039Zc
428AH	0.936AH	0.001IY	0.013OW	0.004N	0.009R	0.001W	0.004xN	0.264Zr	0.049Zc
432AH	0.958AH	0.001IY	0.010OW	0.004N	0.009R	0.001W	0.003xN	0.172Zr	0.057Zc
436AH	0.965AH	0.000IY	0.013OW	0.005N	0.004R	0.002W	0.002xN	0.195Zr	0.059Zc
440AH	0.953AH	0.001IY	0.017OW	0.004N	0.010R	0.001W	0.003xN	0.227Zr	0.051Zc

FIG. 11b

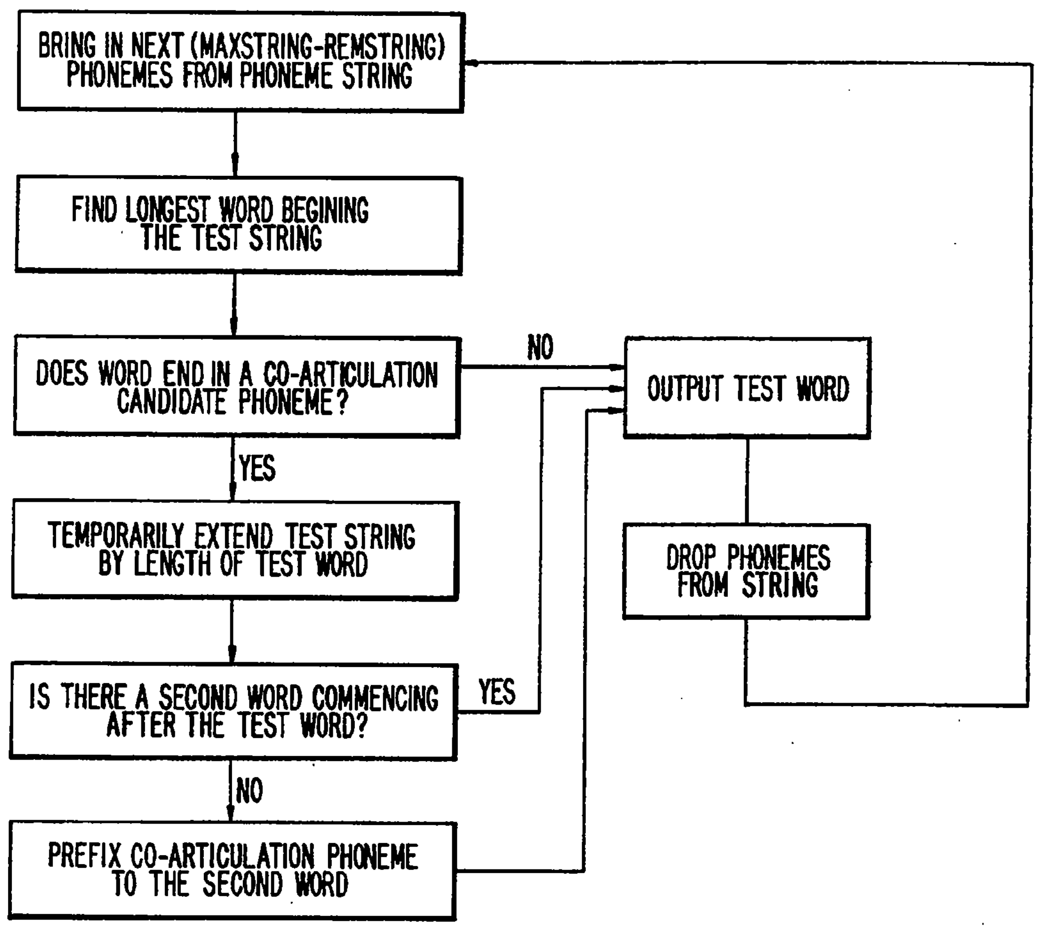
444AH	0.925AH	0.001IY	0.024OW	0.003N	0.014R	0.003W	0.002xN	0.186Zr	0.066Zc
448AH	0.959AH	0.000IY	0.011OW	0.004N	0.012R	0.002W	0.002xN	0.106Zr	0.054Zc
452AH	0.972AH	0.000IY	0.009OW	0.003N	0.004R	0.002W	0.001xN	0.111Zr	0.068Zc
456AH	0.967AH	0.000IY	0.012OW	0.004N	0.007R	0.002W	0.001xN	0.067Zr	0.050Zc
460AH	0.915AH	0.000IY	0.016OW	0.002N	0.012R	0.003W	0.001xN	0.109Zr	0.055Zc
464AH	0.899AH	0.000IY	0.020OW	0.002N	0.015R	0.005W	0.001xN	0.073Zr	0.054Zc
468AH	0.851AH	0.001IY	0.057OW	0.001N	0.009R	0.005W	0.001xN	0.067Zr	0.055Zc
472AH	0.877AH	0.000IY	0.075OW	0.001N	0.006R	0.006W	0.001xN	0.049Zr	0.069Zc
476AH	0.882AH	0.001IY	0.060OW	0.001N	0.006R	0.004W	0.001xN	0.080Zr	0.050Zc
480AH	0.797AH	0.001IY	0.113OW	0.000N	0.004R	0.009W	0.001xN	0.122Zr	0.059Zc
484AH	0.737AH	0.001IY	0.229OW	0.000N	0.002R	0.009W	0.001xN	0.108Zr	0.038Zc
488AH	0.813AH	0.001IY	0.153OW	0.000N	0.002R	0.016W	0.001xN	0.115Zr	0.042Zc
492AH	0.642AH	0.002IY	0.185OW	0.001N	0.004R	0.034W	0.001xN	0.140Zr	0.029Zc
496..	0.458AH	0.001IY	0.200OW	0.001N	0.006R	0.129W	0.001xN	0.082Zr	0.023Zc
500..	0.306AH	0.002IY	0.304OW	0.001N	0.004R	0.200W	0.001xN	0.088Zr	0.019Zc
504..	0.159AH	0.003IY	0.598OW	0.001N	0.002R	0.165W	0.002xN	0.064Zr	0.010Zc
508OW	0.134AH	0.002IY	0.618OW	0.001N	0.004R	0.118W	0.002xN	0.044Zr	0.016Zc
512..	0.128AH	0.003IY	0.570OW	0.001N	0.003R	0.202W	0.002xN	0.023Zr	0.009Zc
516OW	0.094AH	0.005IY	0.791OW	0.000N	0.002R	0.068W	0.004xN	0.011Zr	0.006Zc
520OW	0.243AH	0.005IY	0.629OW	0.001N	0.002R	0.019W	0.006xN	0.011Zr	0.008Zc
524OW	0.133AH	0.009IY	0.771OW	0.002N	0.003R	0.024W	0.011xN	0.006Zr	0.011Zc
528OW	0.167AH	0.009IY	0.743OW	0.002N	0.003R	0.015W	0.014xN	0.004Zr	0.007Zc
532OW	0.215AH	0.011IY	0.621OW	0.003N	0.003R	0.017W	0.017xN	0.004Zr	0.012Zc
536..	0.186AH	0.016IY	0.402OW	0.006N	0.004R	0.058W	0.022xN	0.012Zr	0.015Zc
540..	0.110AH	0.013IY	0.534OW	0.005N	0.005R	0.061W	0.026xN	0.005Zr	0.012Zc
544OW	0.139AH	0.013IY	0.621OW	0.003N	0.005R	0.018W	0.029xN	0.004Zr	0.011Zc
548..	0.162AH	0.015IY	0.480OW	0.004N	0.003R	0.039W	0.025xN	0.006Zr	0.014Zc
552OW	0.127AH	0.013IY	0.637OW	0.004N	0.003R	0.019W	0.030xN	0.003Zr	0.015Zc
556OW	0.075AH	0.011IY	0.837OW	0.003N	0.002R	0.009W	0.036xN	0.001Zr	0.012Zc
560OW	0.083AH	0.010IY	0.884OW	0.002N	0.002R	0.003W	0.040xN	0.001Zr	0.011Zc
564OW	0.135AH	0.009IY	0.908OW	0.001N	0.002R	0.001W	0.037xN	0.000Zr	0.008Zc
568OW	0.149AH	0.008IY	0.932OW	0.001N	0.001R	0.001W	0.056xN	0.000Zr	0.005Zc
572OW	0.110AH	0.008IY	0.950OW	0.002N	0.001R	0.001W	0.060xN	0.000Zr	0.008Zc
576OW	0.068AH	0.007IY	0.967OW	0.001N	0.002R	0.001W	0.077xN	0.000Zr	0.005Zc
580OW	0.074AH	0.006IY	0.958OW	0.002N	0.002R	0.001W	0.080xN	0.000Zr	0.004Zc
584OW	0.120AH	0.008IY	0.912OW	0.006N	0.002R	0.002W	0.091xN	0.000Zr	0.005Zc
588OW	0.108AH	0.009IY	0.922OW	0.008N	0.002R	0.003W	0.081xN	0.001Zr	0.006Zc
592OW	0.066AH	0.007IY	0.952OW	0.007N	0.002R	0.006W	0.060xN	0.001Zr	0.005Zc
596OW	0.078AH	0.006IY	0.932OW	0.010N	0.003R	0.007W	0.058xN	0.001Zr	0.004Zc
600OW	0.108AH	0.005IY	0.892OW	0.017N	0.002R	0.010W	0.066xN	0.001Zr	0.005Zc
604OW	0.119AH	0.005IY	0.903OW	0.015N	0.002R	0.011W	0.050xN	0.001Zr	0.006Zc
608OW	0.080AH	0.006IY	0.950OW	0.007N	0.001R	0.012W	0.053xN	0.000Zr	0.004Zc
612OW	0.113AH	0.006IY	0.925OW	0.005N	0.001R	0.012W	0.047xN	0.001Zr	0.002Zc
616OW	0.158AH	0.005IY	0.866OW	0.007N	0.001R	0.027W	0.043xN	0.001Zr	0.002Zc
620OW	0.182AH	0.005IY	0.814OW	0.006N	0.001R	0.047W	0.032xN	0.002Zr	0.002Zc
624OW	0.116AH	0.005IY	0.891OW	0.002N	0.000R	0.112W	0.018xN	0.001Zr	0.001Zc
628OW	0.087AH	0.004IY	0.860OW	0.002N	0.000R	0.154W	0.024xN	0.001Zr	0.000Zc
632OW	0.105AH	0.003IY	0.865OW	0.002N	0.001R	0.119W	0.028xN	0.000Zr	0.001Zc
636OW	0.068AH	0.003IY	0.906OW	0.003N	0.001R	0.086W	0.046xN	0.000Zr	0.001Zc
640OW	0.065AH	0.002IY	0.928OW	0.002N	0.000R	0.075W	0.040xN	0.000Zr	0.000Zc
644OW	0.031AH	0.001IY	0.927OW	0.002N	0.001R	0.085W	0.037xN	0.000Zr	0.000Zc
648OW	0.038AH	0.001IY	0.914OW	0.002N	0.001R	0.073W	0.061xN	0.000Zr	0.000Zc
652OW	0.037AH	0.001IY	0.861OW	0.004N	0.003R	0.053W	0.102xN	0.000Zr	0.001Zc
656OW	0.039AH	0.001IY	0.882OW	0.003N	0.002R	0.052W	0.101xN	0.000Zr	0.001Zc
660OW	0.032AH	0.001IY	0.863OW	0.004N	0.003R	0.055W	0.098xN	0.000Zr	0.001Zc

FIG. 11c

664OW	0.041AH	0.001IY	0.748OW	0.004N	0.004R	0.119W	0.092xN	0.000Zr	0.001Zc
668OW	0.035AH	0.001IY	0.764OW	0.004N	0.003R	0.159W	0.091xN	0.000Zr	0.001Zc
672OW	0.037AH	0.001IY	0.766OW	0.005N	0.003R	0.156W	0.104xN	0.000Zr	0.001Zc
676OW	0.026AH	0.000IY	0.760OW	0.005N	0.004R	0.121W	0.115xN	0.000Zr	0.001Zc
680..	0.044AH	0.000IY	0.578OW	0.007N	0.007R	0.193W	0.140xN	0.000Zr	0.001Zc
684OW	0.040AH	0.001IY	0.614OW	0.007N	0.005R	0.148W	0.180xN	0.000Zr	0.001Zc
688OW	0.036AH	0.000IY	0.735OW	0.003N	0.003R	0.172W	0.108xN	0.000Zr	0.001Zc
692OW	0.020AH	0.000IY	0.770OW	0.003N	0.004R	0.133W	0.100xN	0.000Zr	0.001Zc
696OW	0.025AH	0.000IY	0.834OW	0.003N	0.004R	0.063W	0.177xN	0.000Zr	0.001Zc
700OW	0.020AH	0.000IY	0.915OW	0.003N	0.003R	0.027W	0.233xN	0.000Zr	0.001Zc
704..	0.016AH	0.001IY	0.956OW	0.003N	0.002R	0.009W	0.436xN	0.000Zr	0.002Zc
708..	0.013AH	0.001IY	0.955OW	0.002N	0.003R	0.004W	0.534xN	0.000Zr	0.003Zc
712..	0.015AH	0.001IY	0.942OW	0.003N	0.004R	0.003W	0.626xN	0.000Zr	0.004Zc
716..	0.007AH	0.001IY	0.943OW	0.003N	0.006R	0.002W	0.656xN	0.000Zr	0.005Zc
720..	0.006AH	0.001IY	0.956OW	0.002N	0.004R	0.001W	0.646xN	0.000Zr	0.008Zc
724..	0.004AH	0.001IY	0.922OW	0.002N	0.004R	0.002W	0.700xN	0.000Zr	0.013Zc
728..	0.006AH	0.001IY	0.821OW	0.004N	0.005R	0.005W	0.538xN	0.000Zr	0.034Zc
732..	0.005AH	0.001IY	0.548OW	0.010N	0.004R	0.039W	0.741xN	0.001Zr	0.045Zc
736..	0.004AH	0.002IY	0.438OW	0.020N	0.002R	0.079W	0.593xN	0.001Zr	0.081Zc
740..	0.005AH	0.002IY	0.360OW	0.021N	0.001R	0.257W	0.547xN	0.001Zr	0.064Zc
744xN	0.004AH	0.001IY	0.299OW	0.024N	0.003R	0.243W	0.631xN	0.002Zr	0.057Zc
748..	0.002AH	0.001IY	0.125OW	0.035N	0.002R	0.426W	0.415xN	0.001Zr	0.089Zc
752..	0.001AH	0.001IY	0.119OW	0.038N	0.002R	0.256W	0.323xN	0.001Zr	0.103Zc
756..	0.001AH	0.000IY	0.011OW	0.104N	0.012R	0.205W	0.068xN	0.002Zr	0.219Zc
760..	0.002AH	0.000IY	0.004OW	0.112N	0.022R	0.187W	0.044xN	0.003Zr	0.180Zc
764..	0.002AH	0.000IY	0.005OW	0.056N	0.048R	0.190W	0.096xN	0.003Zr	0.139Zc
768..	0.001AH	0.000IY	0.004OW	0.057N	0.084R	0.093W	0.073xN	0.002Zr	0.143Zc
772..	0.002AH	0.000IY	0.002OW	0.152N	0.143R	0.039W	0.033xN	0.003Zr	0.114Zc
776..	0.002AH	0.000IY	0.003OW	0.097N	0.335R	0.019W	0.012xN	0.003Zr	0.157Zc
780..	0.002AH	0.000IY	0.002OW	0.073N	0.349R	0.028W	0.007xN	0.003Zr	0.331Zc
784..	0.004AH	0.000IY	0.007OW	0.009N	0.756R	0.020W	0.004xN	0.002Zr	0.341Zc
788..	0.006AH	0.000IY	0.004OW	0.011N	0.647R	0.012W	0.003xN	0.001Zr	0.339Zc
792..	0.007AH	0.000IY	0.010OW	0.008N	0.735R	0.004W	0.005xN	0.001Zr	0.180Zc
796..	0.038AH	0.000IY	0.003OW	0.024N	0.643R	0.002W	0.001xN	0.000Zr	0.171Zc
800..	0.042AH	0.000IY	0.001OW	0.043N	0.452R	0.002W	0.000xN	0.000Zr	0.334Zc
804..	0.063AH	0.000IY	0.001OW	0.040N	0.303R	0.001W	0.000xN	0.000Zr	0.312Zc
808..	0.063AH	0.000IY	0.000OW	0.190N	0.073R	0.002W	0.000xN	0.000Zr	0.416Zc
812..	0.292AH	0.000IY	0.000OW	0.355N	0.086R	0.000W	0.000xN	0.003Zr	0.062Zc
816..	0.139AH	0.000IY	0.000OW	0.338N	0.030R	0.000W	0.000xN	0.012Zr	0.157Zc
820..	0.218AH	0.000IY	0.000OW	0 125N	0.045R	0.000W	0.000xN	0.001Zr	0.168Zc
824..	0.253AH	0.000IY	0.000OW	0.049N	0.127R	0.000W	0.000xN	0.003Zr	0.227Zc
828..	0.332AH	0.000IY	0.000OW	0.072N	0.008R	0.000W	0.000xN	0.002Zr	0.312Zc
832..	0.432AH	0.000IY	0.000OW	0.104N	0.001R	0.000W	0.000xN	0.008Zr	0.224Zc
836..	0.135AH	0.000IY	0.000OW	0.073N	0.000R	0.004W	0.000xN	0.034Zr	0.316Zc
840..	0.158AH	0.000IY	0.000OW	0.053N	0.000R	0.007W	0.000xN	0.025Zr	0.479Zc
844..	0.443AH	0.000IY	0.001OW	0.003N	0.000R	0.011W	0.000xN	0.182Zr	0.661Zc
848..	0.686AH	0.000IY	0.001OW	0.009N	0.001R	0.004W	0.000xN	0.145Zr	0.253Zc
852..	0.593AH	0.000IY	0.002OW	0.010N	0.000R	0.008W	0.001xN	0.157Zr	0.428Zc
856..	0.453AH	0.000IY	0.009OW	0.001N	0.000R	0.010W	0.007xN	0.484Zr	0.531Zc
860..	0.555AH	0.001IY	0.023OW	0.000N	0.000R	0.008W	0.019xN	0.173Zr	0.750Zc
864..	0.294AH	0.002IY	0.014OW	0.001N	0.000R	0.009W	0.025xN	0.444Zr	0.545Zc
868..	0.127AH	0.002IY	0.010OW	0.003N	0.000R	0.015W	0.040xN	0.563Zr	0.509Zc
872..	0.134AH	0.003IY	0.013OW	0.001N	0.000R	0.011W	0.010xN	0.529Zr	0.540Zc
876..	0.145AH	0.001IY	0.008OW	0.002N	0.000R	0.036W	0.015xN	0.589Zr	0.719Zc

FIG. 11d

FIG. 12


BRING IN NEXT (MAXSTRING-REMSTRING) PHONEMES FROM PHONEME STRING
FIND LONGEST WORD BEGINING THE TEST STRING
DOES WORD END IN A CO-ARTICULATION CANDIDATE PHONEME?
NO
YES
OUTPUT TEST WORD
TEMPORARILY EXTEND TEST STRING BY LENGTH OF TEST WORD
DROP PHONEMES FROM STRING
IS THERE A SECOND WORD COMMENCING AFTER THE TEST WORD?
YES
NO
PREFIX CO-ARTICULATION PHONEME TO THE SECOND WORD

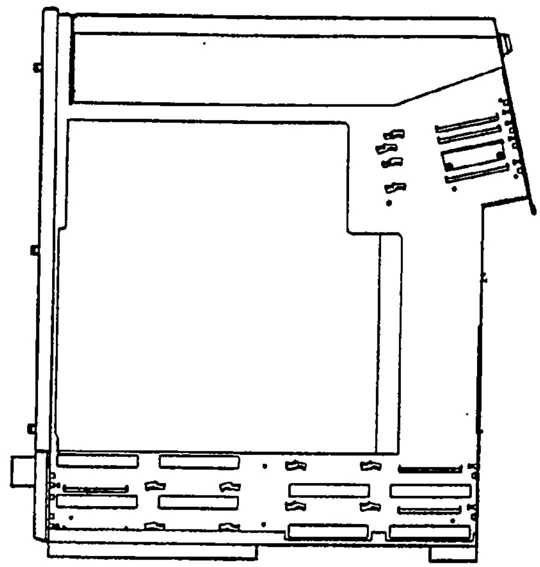


FIG. 13A

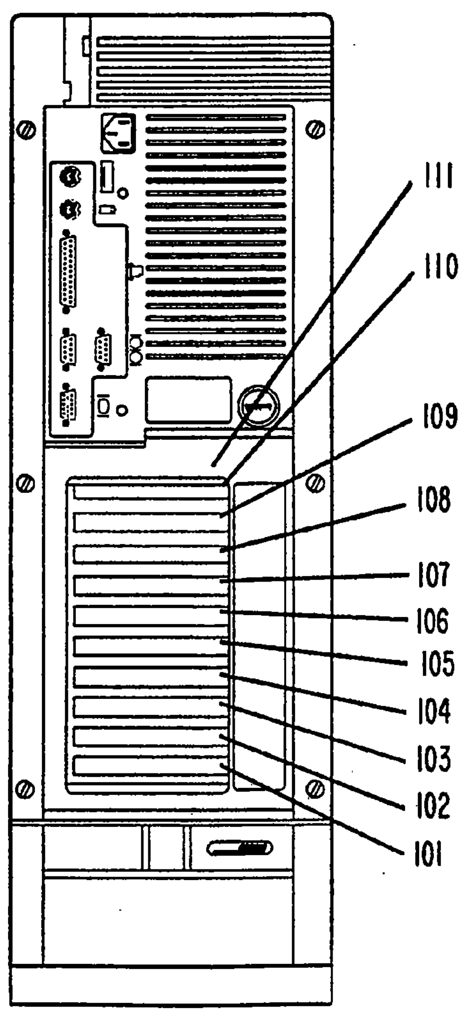


FIG. 13B

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US94/07742

A. CLASSIFICATION OF SUBJECT MATTER

IPC(5) :G 10 L 5/06, 7/08.
US CL :395/2.86, 2.41, 2.44.
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 395/2, 2.86, 2.41, 2.44, 2.5, 2.64-2.66; 381/41, 44.

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

Automated Patent System (APS) search terms:
Speech or Voice, recognition, text, dispay language?, coarticulation.

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y, P	US, A, 5,293,584 (BROWN ET AL) 8 March 1994, see entire document.	1-14
Y, P	US, A, 5,278,911 (BICKERTON) 11 January 1994, see entire document.	1-14
Y	US, A, 4,984,177 (RONDEL ET AL) 8 January 1991, see entire document, especially abstract, figure 2 and col. 9.	1-14
Y	US, A, 4,852,170 (BORDEAUX) 25 July 1989, abstract, figs. 1, 7, 8, columns 1-7 and 9-13.	1-14
Y	US, A, 5,040,215 (AMANO ET AL) 13 August 1991, figs. 6-12, cols. 4, 6--12, especially cols. 11 and 12.	1, 3, 6, 7, 10, 13 and 14.

[X] Further documents are listed in the continuation of Box C. [ ] See patent family annex.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be part of particular relevance

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"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
27 SEPTEMBER 1994	26 OCT 1994

Name and mailing address of the ISA/US	Authorized officer
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INTERNATIONAL SEARCH REPORT

International application No. PCT/US94/07742

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US, A, 4,937,870 (BOSSEMEYER, Jr.) 26 June 1990, see entire document.	1-14.
A	US, A, 4,882,757 (FISHER ET AL) 21 November 1989, see entire document.	1-14.
A	US, A, 4,910,784 (DODDINGTON ET AL) 20 March 1990, see entire document.	1-14.
A	US, A, 5,040,127 (GERSON) 13 August 1991, see entire document.	1-14.
A	US, A, 4,905,285 (ALLEN ET AL) 27 February 1990, see entire document.	1-14.
A	US, A, 4,536,844 (LYON) 20 August 1985, see entire document.	1-14.
A	US, A, 5,033,087 (BAHL ET AL) 16 July 1991, see entire document.	1-14.